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A Balanced Amplifier for Moving Magnet Cartridges

By combining a classic cascode schema with a balanced technique, you can successfully build an outstanding RIAA preamplifier.

By Yury L. Gelgor

Why did I use the word "outstanding" instead of "Hi-Fi" or "Hi-End" in the introductory blurb? Maybe because those fancy words have lost their meanings. "Hi-Fi" now suggests that such equipment is a bit better than the cheapest, and "Hi–End" now assumes the lost meaning of "Hi-Fi."

Well, "Hi–End" is used so often and not always appropriately, so it is already closer to "high tech" or "high priced," rather than about quality of sound. A good friend of mine has the idea of applying the term "Hi–Art" to equipment of outstanding quality. It is probably a good idea; however, no matter what definition I use I am still talking about the same thing: equipment of good sound and musicality.

I did not try to build here a preamplifier with precision RIAA equalization or lowest possible distortions, but I did much to make its sound dynamic, transparent, and musical. Why did I do this now, after I made Retro DAC ("A Balanced Non-Oversampling Retro DAC," June '06, p. 40)? I wanted to compare sound reproductions of CD and vinyl players, and will reveal the results later in the article.

PHOTO 1: Front view.

PHOTO 2: Rear view.

CONNECTING CARTRIDGE TO AMPLIFIER

The input impedance of a vinyl amplifier needs to be 47kΩ and 200-600pF for practically all moving magnet (MM) cartridges. This input capacitance is the sum of several others, such as the capacitance of interconnect cable and input capacitance of the first stage.

Designers suggest that you add an input capacitor when input capacitance is not sufficient; however, this is not always done in practice.

So, how exactly do cartridge manufacturers want you to use their product? According to recommended impedance they do not want you to use it as a pure voltage source, but partially a current one. Let me explain this.

On the one hand, the cartridge electrical equivalent is a serially connected voltage source, resistor, and inductance, and average values of the mentioned resistor and inductance are about 1K and 400mH. Cartridge inductive reactance is low at low frequencies and it increases with increasing frequency. The standard load for the cartridge is 47K; therefore the cartridge impedance is much lower at this value at low frequency and equal or even higher at high frequency. It means at low frequency the cartridge works as a voltage source and at high frequency, when impedances of cartridge and its load are almost equal, it becomes a partial current source.

On the other hand, the cartridge is a reversible electromechanical system, surprisingly similar to any speaker or DC electric motor. It means it produces an electrical signal when the stylus moves and vice versa it produces stylus movement when the signal goes to the coil. Thus, when load impedance is about equal to cartridge output impedance, the current, generated by the cartridge, goes through the load and returns to the coil, affecting stylus movement on mid and high frequencies exactly as low output amplifier's impedance dumping movement of speaker's cone. Every cartridge has its mechanical resonance at high frequency and, I believe, the reason for the suggested connection is to dump this resonance to make high frequency response more flat. The only problem in the suggested connection I see here is the usage not only of a capacitor, but also a resistor, which affects reproduction at mid frequencies. A simple experiment proved this.

I increased input resistor value 20 times and listened to the difference. The advantage of this connection method was significantly improved signal dynamic. Dramatically increased high-frequency level was the disadvantage. The suggested way of connection brings harm to mid frequencies and improves high frequencies, at least to my ear. It needs to be connected another way.

First, I decided to eliminate the input resistor completely by using an input transformer; second, I needed to bring a capacitor to restore the high-frequency load to the input stage. My cartridge now is an Ortofon OMB10 and the value of the input capacitor C is 410pF. Another cartridge may require another capacitor's value, and this is a very good place for the audiophile to play.

HUM, MICROPHONE EFFECT, AND NOISE

A low output signal of the MM cartridge causes several types of distortion: hum, microphone effect, and noise.

Hum is rather easy to handle. DC power supply for the tube heater, proper shielding, and input transformer can reduce it significantly to the level you may not need to worry about it.

Microphone effect is a more serious obstacle. Some tubes have a lower level of it, some have a higher level, and some tubes from the same manufacturer may even have significantly different levels. I'll tell you how to reduce this effect in the construction part of this article.

Noise is not a big deal for tubes there. I cannot guarantee absolute silence when you put your ear against a speaker; however, its level is 15-20dB lower than the noise of an absolutely new record.

INPUT STAGE

There exist several possible input schema configurations.

Pentode schema. High gain, rather low noise level. It is possible to produce a two-stage preamplifier. The main disadvantage is a high level of distortion; it produces not only 2nd and 3rd harmonics, but up to 7th-order. It is also very critical to change load impedance, which sets some limitation on using passive RIAA equalization.

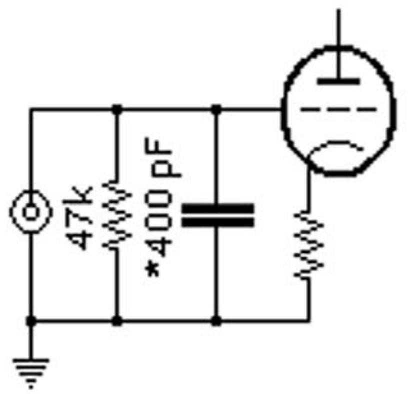


FIGURE 1: Suggested connection.

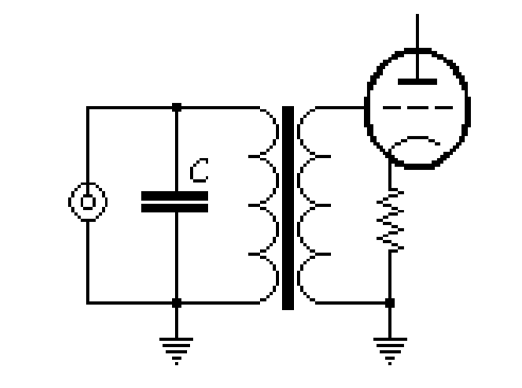


FIGURE 2: Modified connection.

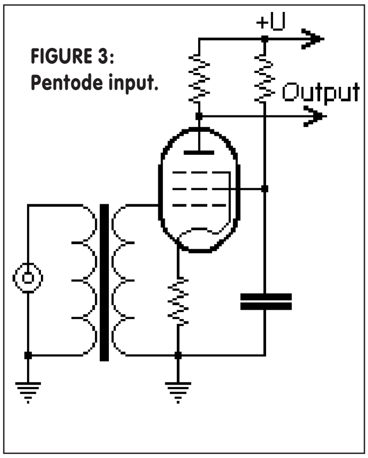


FIGURE 3: Pentode input.

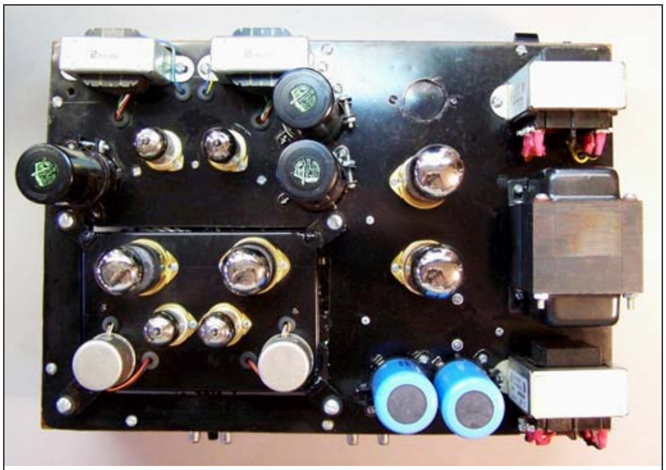


PHOTO 3: Top view.

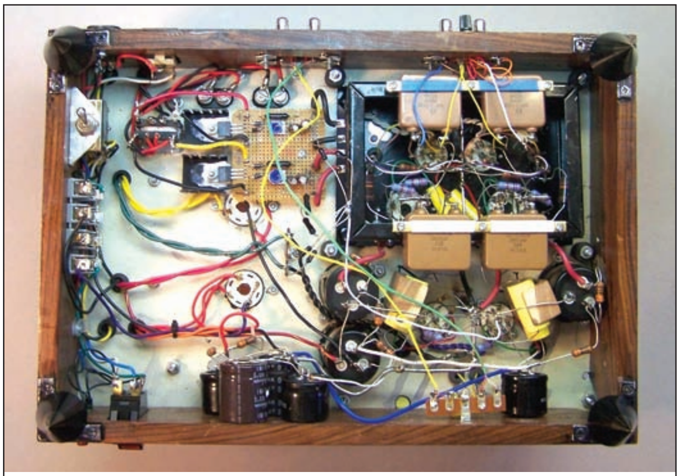


PHOTO 4: Bottom view.

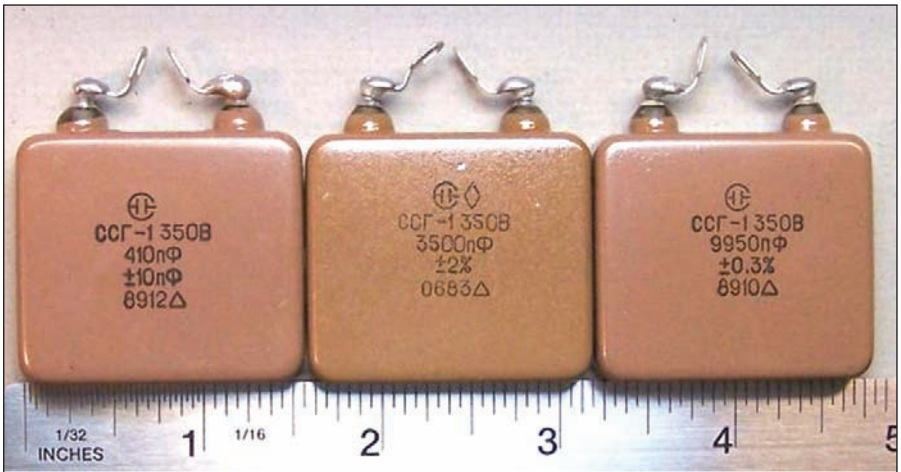


PHOTO 5: SSG-1 410pF, 3500pF, and 9950pF capacitors.

FIGURE 4: Triode input.

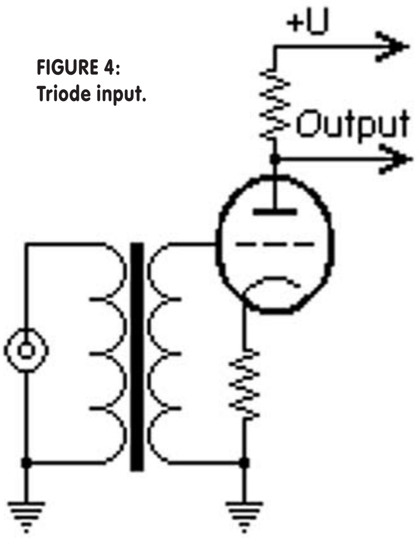


FIGURE 5: SRPP schema.

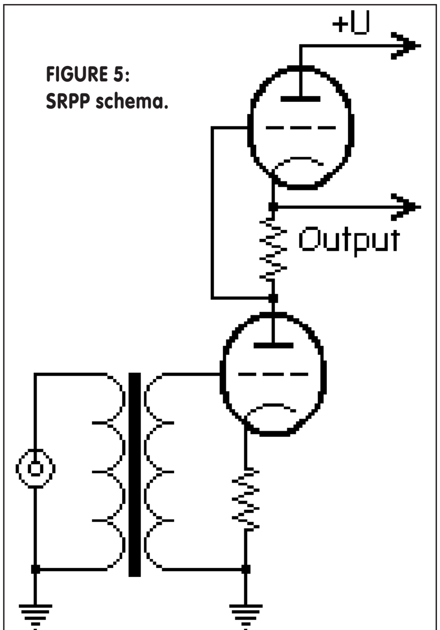
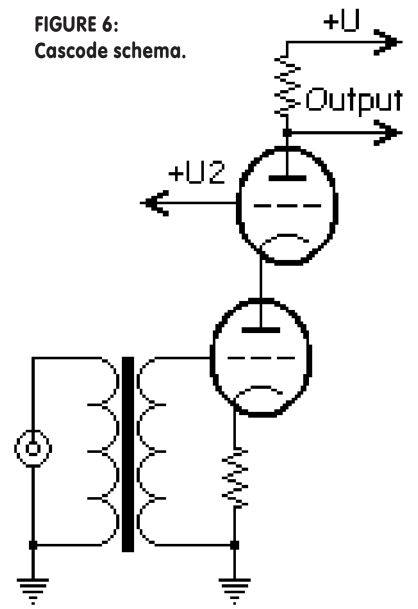


FIGURE 6: Cascode schema.



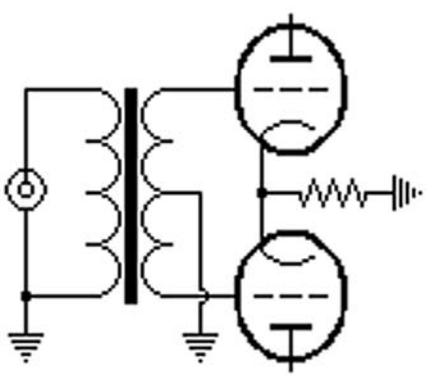


FIGURE 7: Balanced schema and bias.

Triode schema. Almost the best, with low distortion and low noise; however, low gain of this schema requires three stages of amplification, which is obviously not good. It is preferable to have only two.

SRPP (series regulated push-pull) schema. It is very fancy nowadays. Gain is slightly higher than triode. The main advantage is low output impedance, but it offers no benefit for the input stage. The main disadvantage is local negative current feedback.

Cascode schema is my choice. It has pentode gain, triode linearity, and perfect frequency response. High output impedance and the unique property of having good linearity even under low impedance load allows you to use this schema for simplifying RIAA equalization. I'll tell you more about this later.

BIAS AND BALANCED SCHEMA

Realizing bias warrants further discussion. Regular cathode resistors in all of these schemas produce local negative feedback, which reduces gain and also makes any schema sound worse to my ear.

Simply shunting this resistor with a capacitor is actually not good because of the high dependence on the quality of this capacitor. Even the best of Black Gates are not good enough here.

Another idea is to use a ni-cad rechargeable battery instead of the resistor. This is a good arrangement because bias is absolutely equal to battery voltage. But it does not offer any bias adjustment and this is why you should not use this solution.

It is also possible to use a special power supply, but this is one of the worst solutions because sound quality still depends on all parts—capacitors, diodes, regulators, and so forth—of the power supply. It is also more complex and expensive. The only real solution to this problem is a balanced schema.

Direct currents of both rails flow in the same direction adding to each other, producing proper bias on a common-cathode resistor. However, alternative currents flow in opposite directions: input signal in balanced, class A schema increases plate current in one rail and reduces—at the same time—plate current in the other rail for the same amount, therefore the common-cathode resistor voltage drop is not affected by the signal and it means this schema does not produce local negative feedback. Practically, the rails' misbalance is about several percent, and this is a natural way to eliminate negative feedback.

RIAA EQUALIZATION

I used the application of passive RIAA realization from the article "On RIAA Equalization Networks" by Stanley P. Lipshitz[1]. There is nothing new except one thing: I wanted the advantage of using balanced cascode schema.

First, high output impedance of the cascode schema can play the role of resistor R1. Of course, output impedance is not absolutely equal to the calculated value of R1, but it is rather close and the only practical difference is the level of lowest frequencies, which is not that critical. Second, this schema connects between two cascode outputs and the output signal to balanced output stage.

This is the most critical place in the whole preamplifier. The highest quality parts available are needed for this circuit.

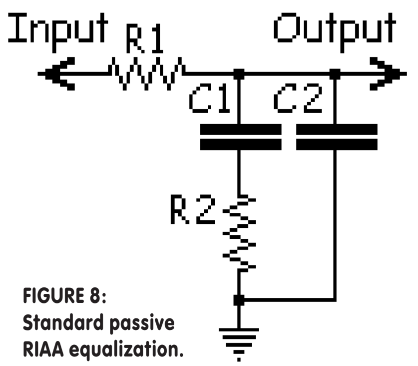


FIGURE 8: Standard passive RIAA equalization.

OUTPUT STAGE

Balanced Schema. This is also nothing new. I used a slightly changed schema of the analog part from my balanced DAC[2]. The difference is that I eliminated the negative power supply and used 6922 tubes. This transformer output is good for production of a 600Ω line to connect to the amplifier.

Preamplifier Schema. I show operating voltages for reference only. You do not need to set the exact voltage value, and 30% tolerance is absolutely fine.

Power supply. There is nothing new in this schema either. The only secret is good quality parts for good sound.

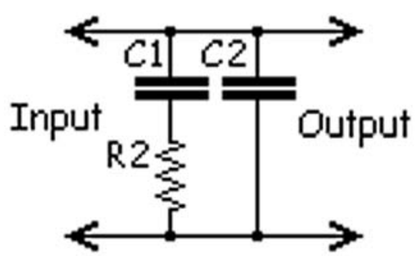


FIGURE 9: Modified passive RIAA.

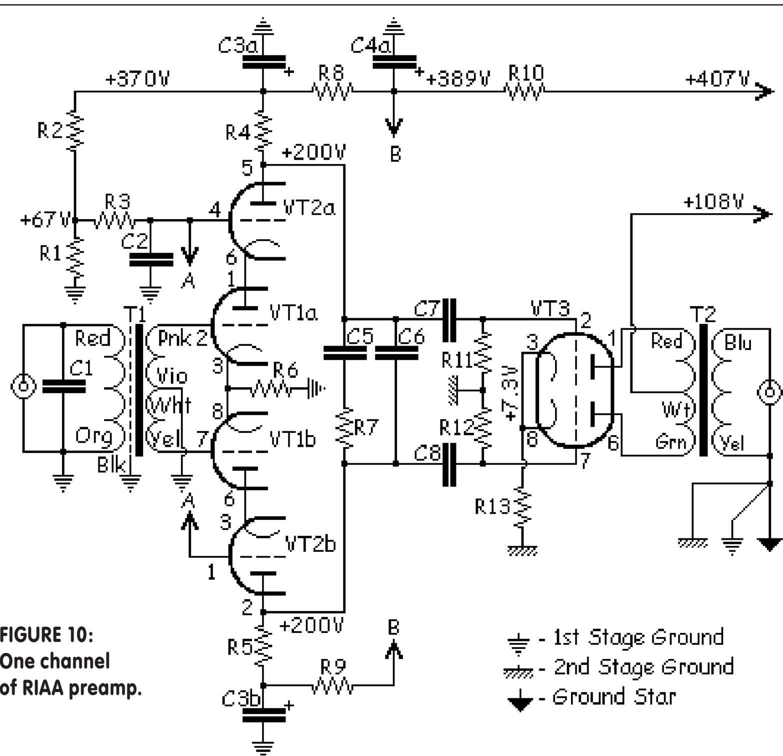


FIGURE 10: One channel of RIAA preamp.

GROUND AND POWER SUPPLY LAYOUT

This part is rarely explained and deserves additional attention.

You connect all first stage ground wires of both channels to a point G1: transformer T1, resistors R1, R6, capacitors C3, C9, C10, and negative 407V power supply. Both resistors R10 and positive 407V power supply go to a point P1. All second stage ground wires go to point G2: resistors R11, R12, R13, secondary T2, and negative 108V power supply. Both taps of the T2 primaries and positive 108V power supply go to point G2.

Then you connect points G1 and G2 to point G3 and only at this point do you connect it to the main chassis. The input stage chassis is electrically connected to the main chassis with only one wire.

CONSTRUCTION

Numbers in parentheses refer to **Fig. 13**. You should make the input stage on a separate chassis (3) and suspended against the main chassis (1) on rubber grommets (4) for vibroisolation. Solder four mounts (2) to the corners of the input stage chassis. Use two grommets on each corner to make this suspension softer. Screw (6) through the grommets and fasten with a nylon insert lock nut (5). I used grommets from laser pickup suspensions.

The rest of construction is similar to my RetroDAC[2]. I used 15″ × 10¼″ steel sheet for the main chassis and put it on a wooden frame, which I glued before and put steel angles on the bottom side of this frame for rigidity. For further vibroisolation I put all construction on speaker spikes. This greatly reduced the possibility of microphone effect.

All wires that go from the main chassis to the input stage chassis must be as soft as possible. Tester probe wire is a good choice. Grounds are made of 8 gauge (about 3.5mm) solid copper wire. All heater wires are made as twisted pairs.

Black Gate capacitors C3, C4, and C106 are pretty tall, about 5″, and may increase the total height of the preamp. To avoid this I drilled holes in the chassis, put caps through, and leveled them with other tubes and capacitors.

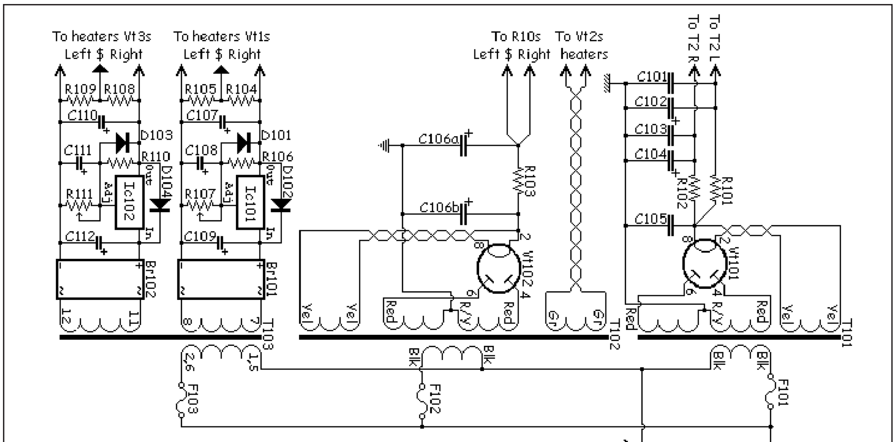


FIGURE 11: Power supply.

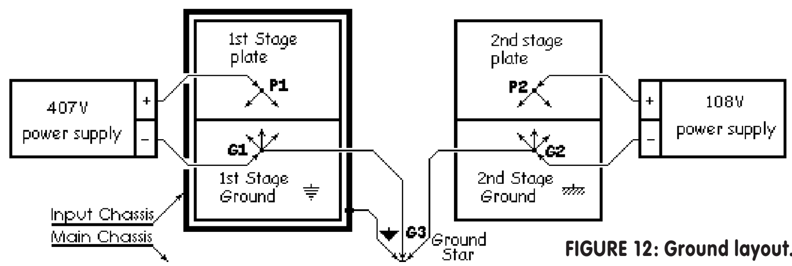


FIGURE 12: Ground layout.

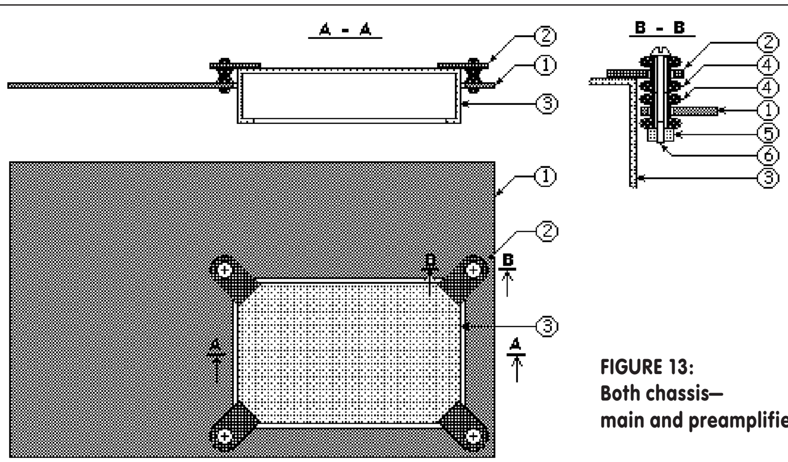


FIGURE 13: Both chassis—main and preamplifier.

STARTUP

The tune-up procedure is simple:

1. Do not connect any interconnect cables. Remove all tubes.
2. Use potentiometers R107 and R111 to set output voltages of tube heater regulators to 6.3V on pins 4 and 5 of tube sockets Vt1 and Vt3.
3. Check all necessary AC voltages.
4. Turn unit off, put in all tubes. Turn on and wait for 10-15 minutes allowing warmup for all tubes.
5. Check heater voltages again on Vt1 and Vt3 and make necessary adjustments with R107 and R111 if needed.
6. Make sure you follow all DC voltages, as shown on the schema. Attention! Vt2a and Vt2b plate voltages cannot be measured with a low impedance voltmeter because of high output impedance of the cascode schema. You must use a high input impedance voltmeter, at least 50mΩ.
7. It is a good idea to keep this unit on for several hours before listening. Electrolytic capacitors need this time for forming. Of course, you can begin to listen immediately, but realize that the sound will be no good at the very beginning but will improve each hour. It took at least 50 hours to fully complete forming the capacitors to my liking. There is no need to keep the unit on for 50 straight hours without turning it off. Forming accumulates and in a week of listening will be completed.
8. Turn off, plug in interconnect cables, turn unit on.

FINE-TUNING

I already mentioned about playing with the value of capacitor C1, which is very important. Now here's what you can do with the RIAA equalization circuit: C5 and C6. Take several capacitors of the same values but different types: 10,000pF and 3300pF or close values. Changing them will not change the frequency response of the unit; however, its sound will change dramatically. There are only three types of capacitors I can recommend: silver mica, tin and Teflon, and copper and paper in oil. Frankly, I did not try silver and paper in oil.

It has become good practice to locate audiophile-grade parts among old Russian military production. This is also the route I plan to follow and want to introduce you to another type of silver mica cap. Their types are SSG-1, SSG-2, and SSG-3, with values from 150pF up to 200,000pF, all rated 350V, and with absolutely stunning tolerance from 2% up to 0.3%!

They look pretty conservative, but the biggest advantage, of course, is the outstanding sound. I compared sound with regular silver mica, Teflon TFT Relcap, and Jensen copper and paper in oil in place of C5 and with no hesitation put SSGs in place of C1, C5, and C6. You can find some of them on eBay. Of course, you may prefer another sound and different capacitor types.

Another place to change sound dramatically with no change in frequency response is to play with plate power supply capacitors C101/C102 and C103/C104. Each of these capacitors has summary capacitance of about 1000µF. Why did I make each of them out of two different capacitors: 820µF Nichicon and 220µF Black Gate? I decided this after much listening.

With just the 1000µF Nichicon cap, the unit sounds as though it is going through curtains. Using just such capacitors affects the sound quality of high frequencies. Black Gates sound perfect, however making a 1000µF battery from just them is an economical disaster. So I decided to mix them up to make it more affordable and get good sound balance at the same time.

Try the different types of capacitors and experience different sounds, some of which you may like. Tastes differ.

HOW DOES IT SOUND?

In the beginning of this article I promised to compare the sound of my DAC and vinyl preamplifier. Here are some surprising results.

First, I used the same CDs and LPs to listen to the same songs on both units. The DAC sounded more transparent, with better stereo separation of channels. Vinyl had better musicality of program and delivery of musical impression. In other words: the DAC produced more precise but cold sound while vinyl sounded more warm and moody.

Use the DAC when you want to hear some background instruments or voices. And use vinyl for listening to the music.

So which one is better? I would say each of them is good. . . in its own way. I am going to use them both, because of their "outstanding quality."

One last word: In my real vinyl preamplifier, I used power-supply schemas exactly as I did before in my RetroDAC[2], with no output stage plate transformer. I just wanted to show you a regular way of doing this and introduce different options. The rest of this vinyl preamplifier is as described here. *aX*

Born in the Ukraine, ***Yury Gelgor*** *received his bachelor's degree in industrial engineering. He is an experienced industrial electronic technician, electronic engineer, and computer programmer. He moved to the US in 1994. He became a hi-fi enthusiast in the early 70s, began to "breadboard" solid-state amplifiers, and has completed about 50 of them. In about 1985, he found a big old radio for a "good old" speaker and since then has fallen in love with tube sound. He dreams of finding a job that matches his hobby.*

REFERENCES

1. "On RIAA Equalization Networks," Stanley P. Lipshitz, *J.* Audio Eng. Soc., 1979, Vol. 27/6.

2. "Completely Balanced Nonoversampling RetroDAC" Y. Gelgor., *audioXpress* 6/06.

TABLE 1: PARTS LIST.

Position	Value	Manufacturer	Supplier
BR101, BR102	5A	Any diode bridge 5A or more	
C1	410pF	Silver Mica	www.newark.com
C2	0.22µF 600V	Hovland Musicap	www.partsconnexion.com
C3, C4	100+100µF 500V	Black Gate WKZ	www.partsconnexion.com
C5	0.01µF 400V	TFT Relcap, Silver Mica	www.partsconnextion.com
C6	3300pF	Silver Mica	www.newark.com
C7, C8	0.22µF 400V	TFT Relcap	www.partsconnexion.com
C101, C103	220µF 160V	Black Gate NH	www.partsconnexion.com
C102, C104	820µF 200V	Nichicon GK	
C105	68µF 350V	Black Gate NH	www.partsconnexion.com
C106	100+100µF 500V	Black Gate WKZ	www.partsconnexion.com
C107, C110	100µF 16V	Any	
C108, C111	1µF 16V	Any	
C109, C112	23000µF 16V	Mallory	www.newark.com
F101, F102, F103	1A 250V	Any slow fuse	
Ic101, Ic102	LM338T	Any	
R1	20k 0.5W	Tantalum	www.partsconnexion.com
R2	100k 2W	Tantalum	www.partsconnexion.com
R3	2M 0.5W	Tantalum	www.partsconnexion.com
R4, R5	120k 1W	Tantalum	www.partsconnexion.com
R6	600Ω 0.5W	Tantalum	www.partsconnexion.com
R7	33k 1W	Tantalum	www.partsconnexion.com
R8-10, R101, R102	3.3k 1W	Tantalum	www.partsconnexion.com
R11, R12	1.0M 0.5W	Tantalum	www.partsconnexion.com
R13, R103	2.2k 1W	Tantalum	www.partsconnexion.com
R104, R105, R108, R109	470Ω 1W	Any	
R106, R110	120Ω 0.5W	Any	
R107, R111	500Ω 1W	Any potentiometer	
T1	8920	Sowter, UK	www.sowter.co.uk
T2	8650	Sowter, UK	www.sowter.co.uk
T101	263AX	Hammond	www.newark.com
T102	272DX	Hammond	www.newark.com
T103	185D16	Hammond	www.newark.com
VT1, VT3	6922		www.partsconnexion.com
VT2	6188		www.partsexpress.com
Vt101, Vt102	5AR4		www.partsexpress.com
Steel Chassis	1441-8	Hammond	www.partsexpress.com
Speaker spikes	DSS4	Dayton	www.partsexpress.com

High-Quality Stepped Attenuators For Less

Back

Tips on factors to consider when building a low-cost stepped attenuator.

By Milan Uskokovic

Let me get one thing straight right away: This is not a "Complete Guide to Secrets of Producing High-Quality Attenuators for One Thin Dime in the Comfort of Your Own Home." My intention here is to show you how to make an attenuator perfectly customized for your application and design requirements, without spending nearly as much as you would on buying that standard stepped stuff from your local retailer. How little money you will actually spend depends on several factors that you need to consider very carefully before you get that soldering iron of yours hot and ready. You must choose the attenuator type, decide on the number of steps you need, select proper resistors for the job, and define the attenuator's nominal resistance.

COMMON TYPES

You can use a serial-type attenuator (**Fig. 1**) as a direct substitute for a common potentiometer. It consists of a chain of resistors connected in series. The number of resistors in the chain, as well as the number of switch steps, depends on the attenuator resolution desired. The main disadvantage of this type of attenuator is that all resistors are in the signal path at one time, each of them detracting somewhat from the quality and overall performance of the attenuator.

FIGURE 1: Serial-type attenuator.

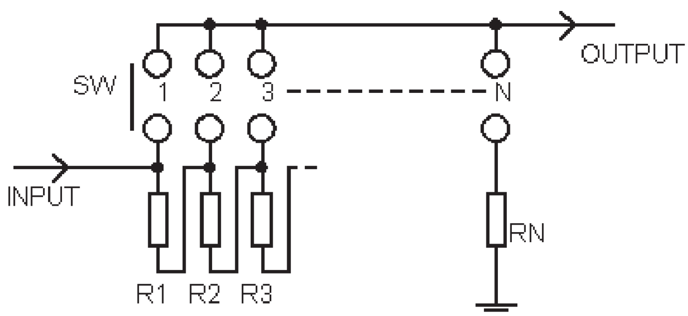
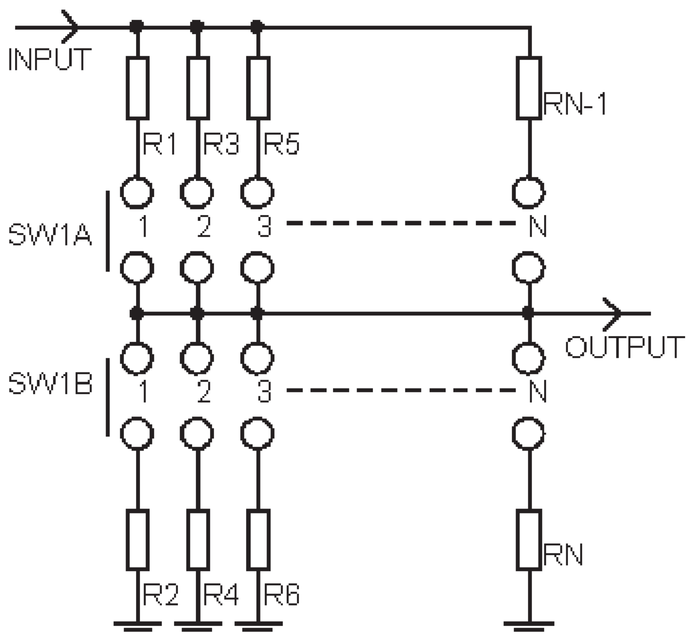


FIGURE 2: Ladder attenuator.



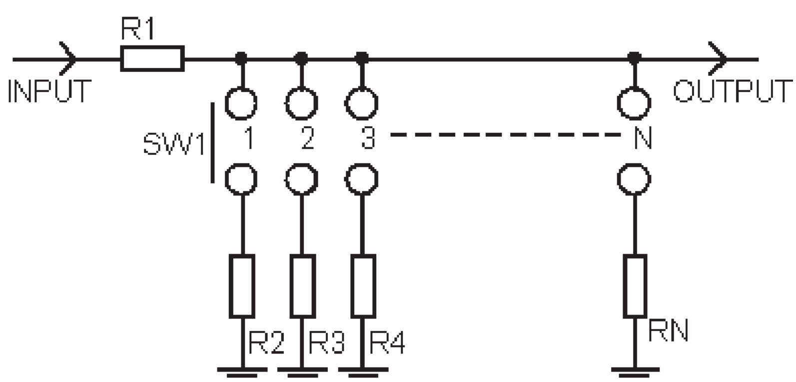
A ladder attenuator (**Fig. 2**) overcomes this major shortcoming of the serial type by utilizing one switch with two resistor layers per channel. Because there are only two resistors in the signal path, resistor noise is negligible. On the other hand, its comparative disadvantages include two switch contacts and twice as many resistors required.

A shunt version (**Fig. 3**) performs like a ladder attenuator, only with fewer resistors. It is not as popular as series and ladder type attenuators, mostly due to its variable input impedance.

NUMBER OF SWITCH STEPS

The number of steps or switch positions depends on the attenuator resolution desired. It has been experimentally established that you need a minimum resolution of 2dB per step in the attenuation range you use the volume control most often. In other words, you do not need fine control across the en-

FIGURE 3: Shunt attenuator.



tire attenuation range, particularly not at the bottom of the volume control range (low volume levels), where attenuation is highest, so you may set the attenuation per step to 3dB or more.

Assuming that the overall attenuation range is 60dB, a 24-position switch should provide a sufficiently precise signal-level control. Attenuators with 12 steps are less commonly used because the steps are too far apart to provide adequate resolution. On the other hand, commercial switches with more than 24 steps (typically 32 or 48) are too expensive to warrant their use in most DIY projects.

WHAT RESISTORS TO USE

The principal factor you need to consider when selecting the right resistors is their precision, which affects attenuation accuracy per step and channel-to-channel signal level tracking. As a general rule, you should use 1% precision resistors for 24-step attenuators, or better if you are building a more complex attenuator design.

Because the signal path passes through resistors, the more resistors you use the higher possible resistor noise in the signal path. However, practical experience shows that resistors with the lowest noise characteristics are not necessarily the best-sounding to our ears, due to individual differences in psychoacoustic aspects of the human hearing system or, simply put, due to our different individual tastes. It is necessary, therefore, that you try different brands available in the market and stick to the one you believe to be best suited to your sonic preferences and application.

Cost-wise, better-sounding resistors are usually more expensive. For instance, standard 1% metal-film resistors can cost as little as $0.1 apiece while, on the other hand, tantalum resistors typically cost up to $5 apiece, which can obviously make quite a huge difference and add up to significantly more than your budget will stand.

ATTENUATOR'S NOMINAL RESISTANCE

In my previous article ("Load Impedance and Stepped Attenuators," June

'05), I discussed the effect of the input impedance of amplifiers/preamplifiers on the resolution of your attenuators. The input impedance of an attenuator will also depend on the source signal. As a rule of thumb, the nominal resistance of the attenuator should be between 10 and 100kΩ. This is not critical where the input impedance of the signal source is low (100Ω) and that of the amplifier is high (100kΩ).

As regards induced noise levels and frequency linearity, it is always better to use the lowest possible value, whereas the highest possible value will make the load easy on the source.

CHOOSING THE RIGHT ATTENUATOR

If you tend to play your audio system at a limited range of volume levels (e.g., moderate), you do not really need a volume potentiometer with 24 positions. In that case you may wish to consider building an attenuator with fewer steps and a 2dB per step attenuation curve, in which you would be able to fine-tune the steps to the sound levels that cater best to your listening preferences. For most people, that range is typically between −20 and −10dB, which is where your attenuator should perform best.

Outside that attenuation range, when playing at low volume levels and you do not want your music to be too intrusive (e.g., soft background music), cheaper resistors will do just fine. Besides, the step from active to passive listening is a relatively big one—20dB or more. For that reason, broadcast consoles have a special dimmer switch that attenuates the signal by 20dB so you do not need to use an attenuator at all.

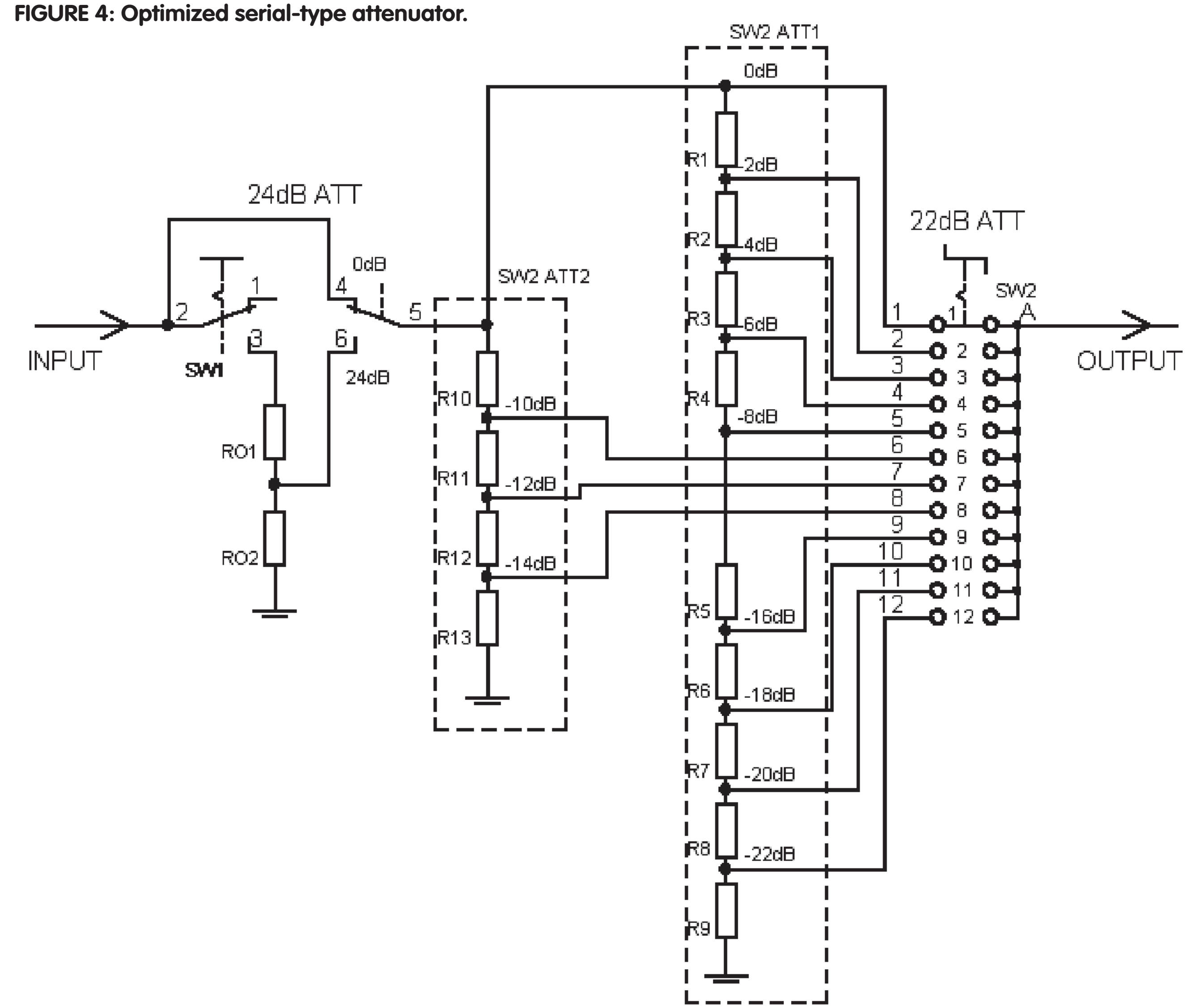
In the next part of this article, I will show you how to build and optimize a good-quality serial-type attenuator with dimmer function and a 12-step make-before-break switch.

IMPLEMENTATION

One such implementation is shown in **Fig. 4**. SW1 is a two-pole switch (for two channels, you will need a four-pole one) with two positions that allow you to choose an attenuation value between 0 and 24dB. It can be either a push-button or toggle switch, and it functions as a dimmer.

SW2 is a one-pole (for stereo, you should use a two-pole unit), 12-position make-before-break-type rotary switch. It allows you to adjust the volume level in 12 steps of 2dB each, providing an at-

FIGURE 4: Optimized serial-type attenuator.



tenuation range of 0dB to –22dB when SW1 is set to 0dB. When you set SW1 to –24dB, the attenuation range of SW2 will be between –24 and –46dB. Thus, you achieve a total attenuation range of 46dB in 2dB steps.

A first glance reveals a somewhat atypical implementation of the series attenuator SW2 resistor network. The attenuator, in fact, consists of two series attenuators—SW2 ATT1 and SW2 ATT2. SW2 ATT1 circuitry comprises resistors R1 through R8, covering the attenuation range from 0 to –8dB and from –16 to –22dB. SW ATT2 is made up of resistors R10, R11, R12, and R13, with the attenuation range from -10 to –14dB. The total attenuation range is thus split into two segments—the comparatively large range, which is used less often (SW2 ATT1), and the smaller but, I assume, the most frequently used attenuation range (SW2 ATT2).

Here is a money-saving tip for you frugal DIYers: Do not throw away your hard-earned cash on high-quality resistors for the attenuation range, which is of only secondary importance for your

FIGURE 5: Spreadsheet results for SW1.

	B	C	D	E	F	G	H	I	J	K
1										
2	LOADED 24-STEP ATTENUATOR CALCULATOR									
3		Ratt(kOhm)		Rina(kOhm)	Source Zout(kOhm)					
4		11		11,00		0				
5										
6										
7	Step	Sattn(dB)	Attn(k)	Rx		Rn(kOhm)		Rin(kOhm)	SinAtt(dB)	
8	1	0	1,00	11,000	R1=	10,263		5,50	0,00	
9	2	24	15,85	0,737	R2=	0,737		10,95	0,00	
10	3	100	100000,00	0,000	R3=	0,000		11,00	0,00	
11	4	100	100000,00	0,000	R4=	0,000		11,00	0,00	

FIGURE 6: Spreadsheet results for SW2 ATT1.

	B	C	D	E	F	G	H	I	J	K
1										
2	LOADED 24-STEP ATTENUATOR CALCULATOR									
3		Ratt(kOhm)		Rina(kOhm)	Source Zout(kOhm)					
4		22		100,00		0				
5										
6										
7	Step	Sattn(dB)	Attn(k)	Rx		Rn(kOhm)		Rin(kOhm)	SinAtt(dB)	
8	1	0	1,00	22,000	R1=	3,958		18,03	0,00	
9	2	2	1,26	18,042	R2=	3,478		19,24	0,00	
10	3	4	1,58	14,564	R3=	2,934		20,15	0,00	
11	4	6	2,00	11,631	R4=	2,403		20,79	0,00	
12	5	8	2,51	9,228	R5=	5,636		21,22	0,00	
13	6	16	6,31	3,592	R6=	0,753		21,88	0,00	
14	7	18	7,94	2,838	R7=	0,594		21,92	0,00	
15	8	20	10,00	2,244	R8=	0,468		21,95	0,00	
16	9	22	12,59	1,776	R9=	1,776		21,97	0,00	
17	10	100	100000,00	0,000	R10=	0,000		22,00	0,00	
18	11	100	100000,00	0,000	R11=	0,000		22,00	0,00	

FIGURE 7: Spreadsheet results for SW2 ATT2.

	B	C	D	E	F	G	H	I	J	K
1										
2	LOADED 24-STEP ATTENUATOR CALCULATOR									
3		Ratt(kOhm)		Rina(kOhm)	Source Zout(kOhm)					
4		22		100,00			0			
5										
6										
7	Step	Sattn(dB)	Attn(k)	Rx		Rn(kOhm)		Rin(kOhm)	SinAtt(dB)	
8	1	0	1,00	22,000	R1=	14,704		18,03	0,00	
9	2	10	3,16	7,296	R2=	1,535		21,50	0,00	
10	3	12	3,98	5,761	R3=	1,213		21,69	0,00	
11	4	14	5,01	4,548	R4=	4,548		21,80	0,00	
12	5	100	100000,00	0,000	R5=	0,000		22,00	0,00	
13	6	100	100000,00	0,000	R6=	0,000		22,00	0,00	
14	7	100	100000,00	0,000	R7=	0,000		22,00	0,00	

purposes (i.e., SW2 ATT1), when you may do just as well with standard and cheap metal-film ones. SW2 ATT2 used here is quite similar to ladder-type attenuators; there are two switches and only four resistors in the signal path for the three most frequently used attenuation steps!

CALCULATING THE VALUE OF INDIVIDUAL RESISTORS

Start by defining the attenuator's nominal resistance. In my example, where the output resistance of the signal source is low (up to 100Ω) and the input impedance of the amplifier exceeds 100kΩ, the attenuator's nominal resistance is 11k. Naturally, you may use any other values you wish for your attenuators.

Calculate the value of each resistor by using the spreadsheet published online at www.audioXpress.com. First, calculate the values of resistors R01 and R02 for the fixed attenuator, which is loaded with SW2 attenuator whose nominal resistance is R = 11kΩ. Enter the value of 11kΩ in the Rina field. In the attenuation field, enter only 24dB. The figures you get for R1 and R2 are the values of your resistors R01 and R02 (**Fig. 5**).

Now calculate the values of each resistor in the first attenuator (SW2 ATT1). Its nominal resistance is 22kΩ (the two attenuators are connected in

PHOTO 1: Low-power amplifier with attenuator.

PHOTO 2: Finished attenuator.

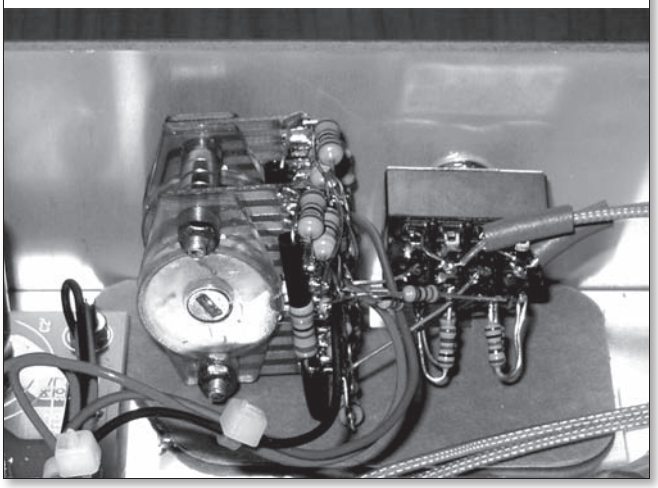


PHOTO 3: Amplifier side view.

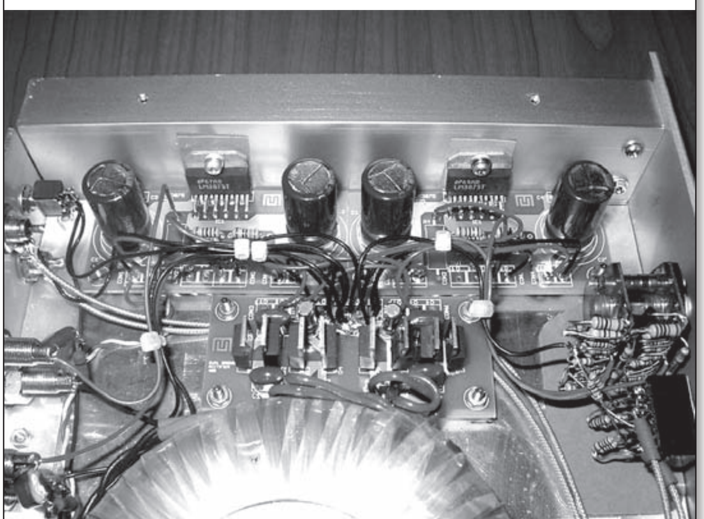
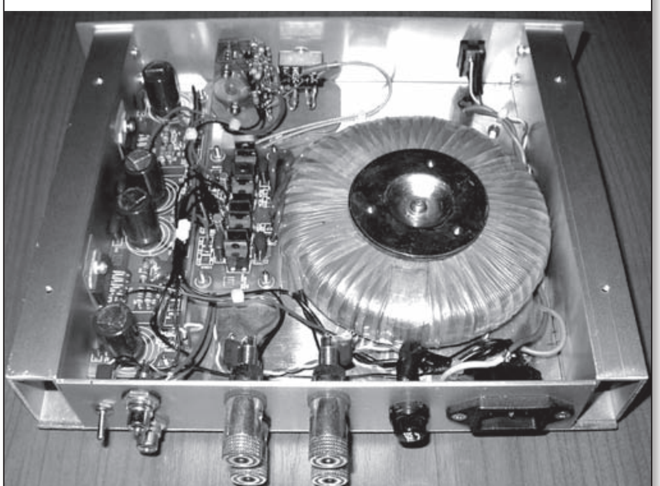


PHOTO 4: Amplifier rear view.



parallel), Rina is 100kΩ or more, and the attenuation step values are 0, 2, 4, 6, 8, 16, 18, 20, and 22dB. The resulting figures are the values of resistors R1 through R9 (**Fig. 6**). Repeat the same calculation procedure to find the values of resistors R10 through R13, but this time enter the attenuation step values of 10, 12, and 14dB (**Fig. 7**) in the table.

Of course, you may make whatever modifications and adjustments you require for your particular project. Thus, you may choose a different nominal resistance for your attenuator, a different per-step resolution, a different attenuation range where you want your attenuator to perform at best (i.e., the one requiring more expensive resistors), and so on. To obtain a resolution of 2.5dB per step, for example, you will need an attenuator with a maximum attenuation of 23 × 2.5 = 57.5dB and SW1 attenuation of 12 × 2.5dB = 30dB.

PRACTICAL EXAMPLE

Photo 1 shows a practical implementation of the attenuator in a low-power IC-based amplifier using the IC LM3875. The resistors in the amplifier circuitry and that part of the attenuator which is most frequently in use during operation are tantalum ones, while those used in the rest of the attenuator are standard 1% metal-film resistors. SW1 is a standard four-pole toggle switch with gold-plated contacts, while SW2 is an average-quality two-pole make-before-break rotary switch with 12 positions (**Photo 2**).

CONCLUSION

One of the major advantages of any DIY project is that you can almost entirely customize it to meet your specific needs and requirements, allowing you to achieve a much more favorable cost-benefit ratio than would be the case if such an optimization were not done. The point of this article has been to show you precisely how to build an optimized DIY attenuator at a price that suits your budget but still fits all your specifications and does the job effectively, without compromising either the quality or usefulness of your project. *aX*

PHOTO 5: ECC86-based battery-powered preamplifier and amplifier with attenuator.

Auri: A Double Chamber Reflex Speaker

"Sometimes They Come Back. . . Again." This is the title of a 1996 horror movie [sequel]. In this article there's nothing scary, just an old reflex variation that reappears.

By Claudio Negro

Auri is a floor-standing two-way speaker that uses the unusual double chamber reflex (DCR) enclosure. This kind of system was proposed in December 1961 in *Electronics World* by G.L. Augspurger and later on resumed first by R. N. Marsh (*Speaker Builder* 3/80) and then by David B. Weems (*Speaker Builder* 4/85 and 1/92 and in several of his books). To avoid confusion, **Fig. 1** shows the bass reflex and four of its variations, including the one of Augspurger-Weems. All measurements were done using the freeware program *Speaker Workshop*[1], with the exception of the room response, and all the SPL values are not absolute.

INSIDE THE DCR

The speaker is made of two chambers, one of which is twice the size of the other, with the woofer placed in the bigger chamber (**Fig. 2**). In the first chamber, V_1, there are two ports: the first, P_1, tunes V_1 with the outside while the second port, P_2, tunes the two chambers, $V_1 - V_2$. Finally, the third port, P_3, tunes V_2 with the outside. Therefore, three different resonance frequencies will be produced: F_1 related to $V_1 - P_1$; F_2 related to $V_1/V_2 - P_2$; F_3 related to $V_2 - P_3$. Keeping $\mathbf{V_1 = 2 \times V_2}$ and the three ports equal to each other, in length and diameter, results in $F_2 = 1.126 \times F_1$, *while* $F_3 = 2.17 \times F_1 = 1.93 \times F_2$.

In the impedance response of a DCR system, there are three peaks instead of the typical two of the bass reflex. You can also see two F_B frequencies: in the lower one (F_2) the woofer sees just one chamber ($V_1 + V_2$), as if there isn't any partition between the two chambers; the other one (F_3) is present at a little bit less than one octave of F_2, with $F_3 = 1.93 \times F_2$. In the DCR impedance chart (**Fig.**

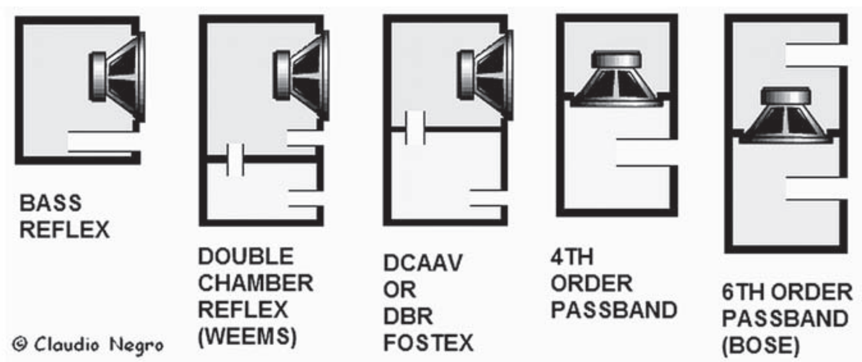


FIGURE 1: The bass reflex and some of its variations.

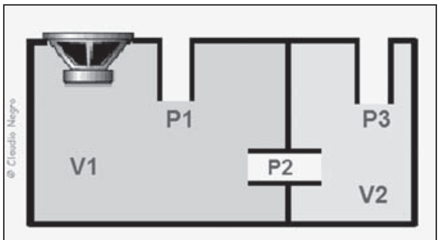


FIGURE 2: DCR chambers and ports.

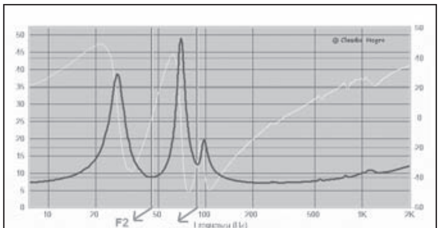


FIGURE 3: DCR impedance response.

3), I found that F_2 = 46.1Hz and F_3 = 89Hz, measured looking at the two dips where the impedance module assumes the relative minimum. The measured values of F_2 and F_3 are confirmed by applying said formulas.

To prove that the DCR F_2 is equal to the F_B of the corresponding reflex system, thus taking off the separation between the two chambers, I compared their impedances. Neither system used an absorbing mat, and the reflex total volume was 0.8 ltr bigger than the DCR one, because of the absence of the chamber's separation panel. Notice in **Fig. 4** how similar the dips between the peaks are, just to confirm that the F_2 of the DCR is equal to the reflex F_B; that is, at F_2 the DCR sees just one chamber ($V_1 + V_2$), simplifying its design.

The typical frequency response of a DCR speaker has a dip just above F_3. But, according to Augspurger and Weems, you needn't worry about this because of its modest amplitude and also because dips are less audible than peaks. I agree with this statement, because you need a well-trained ear to notice what is missing. In case the dip is very deep, Weems suggests you place some absorbing mat inside P_2; however, doing so will increase the port losses altering the known parameters in a not easily predictable way.

The chart in **Fig. 5** is the near field response of just the woofer, thus without having measured and added the ports responses, and it's valid from 20Hz. F_2 is recognized by the big notch, because in a reflex system the woofer less energetic emission happens at its F_B. In this chart you notice that F_2 is equal to 44.5Hz, while in the impedance chart it is equal to 46.1Hz. This difference is normal because usually acoustic data and electric (impedance) data do not agree exactly.

However, I suggest you use the woofer near field notch as the more reliable

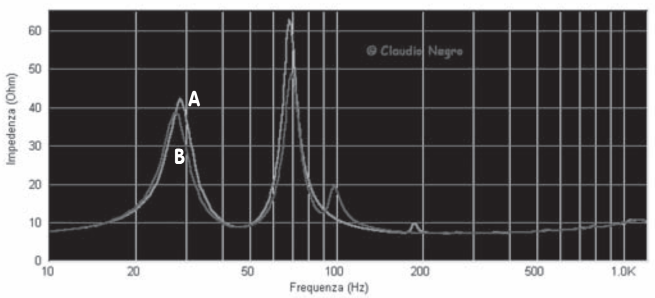


FIGURE 4: Impedance response comparison: REFLEX (A), DCR (B).

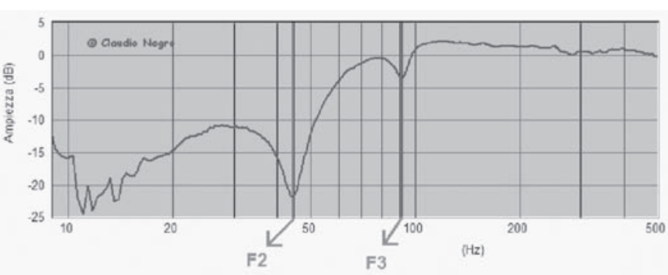


FIGURE 5: DCR near-field response without the ports.

measurement of the reflex tuning frequency. Notice the typical double chamber reflex dip near F_3. But what causes it? To understand why it happens, I studied the two external port acoustic emissions, P_1– P_3, looking at the near field phase chart of P_1– P_3 and of the woofer (**Fig. 6**). Below F_2 the ports are in phase with each other but out of phase with the woofer; between F_2– F_3 the ports are in phase with each other and with the woofer as well; over F_3 the ports are out of phase with each other, which is what causes the DCR typical dip.

The advantage of the DCR compared to a conventional reflex system—as reported by D. B. Keele—is a reduction of the cone excursion around F_3, with a consequent reduction of the distortion. In fact, in a reflex speaker the cone excursion is minimal at F_B (while in a closed box the cone excursion is maximal near the F_C); therefore, the diaphragm nonlinearity distortion, as well as the intermodulation one, is less than in a closed box at the system tuned frequency.

In a DCR the reduced cone excursion happens at two frequencies, F_2 and F_3, and in a wider range compared to a reflex system. Moreover, the use of two chambers in a tower speaker, where it is easier to produce standing waves, helps to reduce the phenomenon of the resonances. In fact, in **Fig. 4**, where I compared the impedance of the DCR with a reflex box obtained by eliminating the chamber's division panel, the reflex peak at 180Hz, due to the box height resonance, disappears.

DCR DESIGN

To start, select a reflex alignment for the driver in use, taking into account all the known procedures to obtain a good-sounding reflex system. To help you decide, you can use one of the numerous software programs, such as the freeware *Unibox*[2]. Then divide the total volume of the reflex (V_B) to obtain a partition that divides the box into two unequal chambers, with one twice as big as the other. You mount the woofer in the bigger chamber. For example, if your reflex V_B is 60 ltr, divide it by 3 to get the smaller chamber volume (V_2 = 20 ltr), which you multiply by 2 to get the bigger chamber volume ($V_1 = 20 \times 2 = 40$ ltr). As you have seen before in the impedance comparison chart, you can assume that $\mathbf{F_B = F_2}$; therefore, you can go on calculating F_1 and F_3 using the known formulas.

But what happens when the chamber volumes are not kept in the 2:1 proportion? I investigated this by leaving the V_2 volume the same while increasing V_1 by one liter first (A) and then decreasing V_1 by two liters (C), in comparison to the ideal 2:1 volumes ratio (B). The DCR total volume ($V_1 + V_2$) was 35 ltr. In the impedance chart in **Fig. 7** you can see the result: While the F_2 is almost the same for the three curves, the F_3 shows a notable difference in the smaller-than-ideal curve.

To continue the chamber ratio change study, I compared the near field response of the three systems in the impedance comparison chart. I measured without adding the port responses because I wanted to see the system tuning frequency ($F_B = F_2$) given by the typical response notch. The near field chart of **Fig. 8** confirms what you already saw in the impedance comparison: The three curves show a F_2 difference inside 0.8Hz; after 70Hz A (one ltr bigger V_1 chamber) and B (ideal 2:1 ratio) curves are similar in shape, while C (two ltr smaller V_1) starts falling down at 83Hz, drawing a deeper and wider dip, for sure more audible. My conclusion is that you should never keep the volume of V_1 less

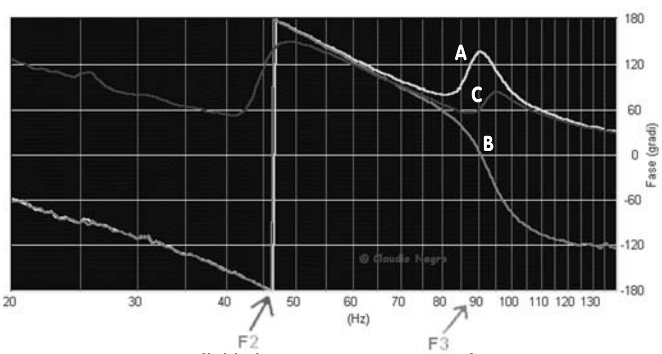


FIGURE 6: DCR near-field phase: P1 (A), P3 (B), woofer (C).

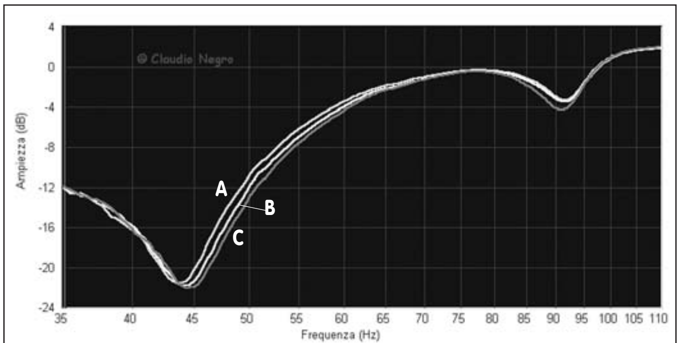


FIGURE 8: DCR near-field comparison with V1: bigger (A), ideal (B), smaller (C).

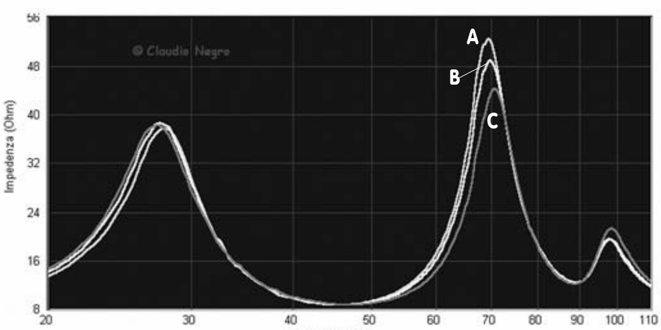


FIGURE 7: DCR impedance comparison with V1: bigger (A), ideal (B), smaller (C).

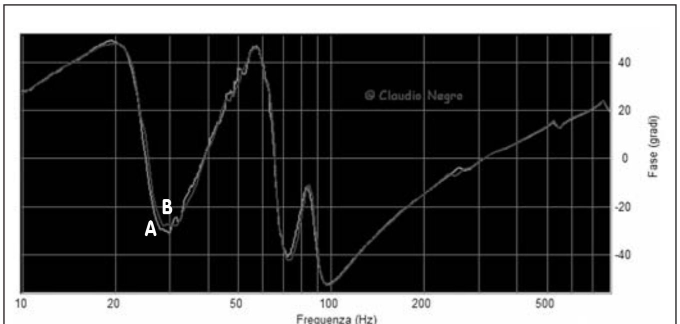


FIGURE 9: DCR impedance phase: P2 tube inside V1 (A), in the middle (B).

than $2 \times V_2$, as it can cause a more evident typical DCR dip; even using $V_1 > 2xV_2$ (inside a 5% increase) doesn't produce problems, just a lower F_2 in comparison to the ideal 2:1 response.

To calculate the port dimensions, knowing the chamber's volume and the desired F_2, just use the volume of the larger chamber (V_1) that tunes at F_1 using the known port calculation formula or a software program. Remember that $F_2 = 1.126 \times F_1$, therefore $F_1 = F_2 / 1.126$. Then use the resulting port dimension for all three ports, so that they are all identical.

Another issue I studied is the position of P_2. Should the tube be inside V_1 or V_2 or in the middle of the two? I measured the impedance phase of the DCR first with the P_2 tube inside V_1 (A) and after with it in the middle of the two chambers (B). As you can see in **Fig. 9**, F_2 is the same for both curves, meaning that the tube position is irrelevant. However, the P_2 tube inside V_1 phase (A) shows more spikes, certainly caused by its closer distance to the P_1 tube. Therefore, in this case, it's better to place P_2 in the middle of the two chambers.

AURI DRIVERS

I intended to use a ribbon tweeter because I like to experiment with uncommon drivers, knowing their sound is rich with details but with a limited vertical dispersion. For some people (mainly ribbon driver resellers), a limited vertical dispersion is a positive feature because it makes the driver less dependent on the listening room. On the other hand, some people maintain that using ribbons produces only one narrow listening point, making group listening difficult.

My thought is that room reflections and reverberations have a big role, explaining why the same speaker might sound good in a room but better in another. One of the advantages of building a speaker is that you can fine-tune it, through the crossover, for your listening room.

But back to my tweeter choice, I ended up with the ATD DDRT120, which is identical to the Silver Flute YAG20-1. Unfortunately for my wallet, the price paid for the ATD in Italy is very different from the Silver Flute sold in the US! It often happens that in my country cheap drivers become hi-end (in price) ones. Anyway, in a two-way system it is important to use a tweeter with a low resonance frequency because its crossover cutoff frequency should be at least one octave above its Fs: the ATD Fs is at 1100Hz, making it a good choice. The woofer, however, needs an extended off-axis frequency response.

Ideally you should use each driver until it works like a rigid piston that can be easily calculated through the formula $F = 345/2d$, where d is the driver effective diameter expressed in meters. However, you must consider the cone material, which can perform miracles in increasing the driver rigidity, and thus its rigid piston frequency range. In fact, the Focal 7K 4412 (my woofer choice), whose cone is made of polykevlar, showed a very good off-axis response allowing a high cutoff frequency; moreover, it has the right Qts for a vented enclosure and a 90dB sensibility. Its measured Thiele/Small parameters are: Fs 44.9Hz, Qts 0.42, Qms 6.99, Qes 0.44, Re 6.06Ω, Vas

37.6 ltr, and Sd 176.7cm^2.

The 7K 4412 frequency response shows a typical rigid cone peak at its break-up frequency, around 2800Hz. I also measured a dip around 1100Hz that wasn't present in the Focal datasheet. I decided to investigate it, noting that in the woofer free air impedance response there was a peak at that frequency, to confirm the reliability of my measurement. So I contacted Focal in France to ask how a dip can disappear like magic.

They confirmed that the dip at 1100Hz really exists but is not visible in the datasheet because their measurement methodology (on infinite baffle) tends to rise up to the cone break-up point, somehow compensating for the dip at that particular frequency. They believe the dip is due to a slight mechanical termination mismatch between the polykevlar cone and the surround.

CABINET

Auri is a floor-standing tower speaker whose internal dimensions (959 × 188 × 218mm, H × W × D) are not in line with the golden ratio. Here's what I've done to try to make a deaf cabinet. All panels are 2cm thick MDF with the exception of the front and rear ones, whose thickness is 3cm. An internal frame is present all around the perimeter, which helps to harden the structure and at the same time serve as an end guide for the front and rear panels. The top, bottom, and lateral panels use L joints (**Photos 1-2**). The front panel has a step (**Photo 3**), so that it's partially stuck inside the cabinet, while its edges are rounded to reduce *cabinet edge diffraction.*

There are actually two schools of thought about how audible cabinet diffraction is. The first affirms that the change from 4π str to 2π str causes an image degradation, and some suggest the use of a compensation circuit. The other states that the room response compensates for any low frequency pressure drop (baffle step or diffraction loss) noticeable in the anechoic response. Moreover, the diffraction effects are very directive while, usually, the listening is made off-axis. I tend to take the middle ground, trying to minimize the diffractions, since eliminating them completely is impossible, by using a narrow front panel that contributes toward a better sound image, and by rounding its edges with a radius of at least 1″, as reported by J. D'Appolito and J. Moriyasu[3].

I also flush-mounted all the drivers to reduce diffraction at the driver's edge, and enlarged the woofer internal hole because the panel thickness (3cm) could have produced an unwanted resonance (**Photos 4-5**). On all internal sides, excluding the front one, I used a product called *Fonogel* from AZ Audiocomp, which is a viscous-elastic damping compound that's non-toxic and doesn't smell like solvent-based products. I also used a damping mat on all panels except the front one, choosing from the long fiber polyester and the rusticated expanded polyurethane. Each type comes with a damping factor chart. So knowing the cabinet internal dimensions and applying the formula $F = 345/2d$, where d is expressed in meters, I was able to use the better mat type for that particular resonance frequency.

Another issue I addressed is the transmission of vibrations from the woofer to the cabinet and the tweeter. First of all, I must say that various modes of panel vibration, or resonance, exist. The first mode is referred to as forward-backward panel movement; the second mode occurs when a half panel moves forward while the other half moves backward, divided on the vertical axis; the third mode is like the second one but on the

PHOTO 1: Cabinet internal frame.

PHOTO 2: Panels L joints.

PHOTO 3: Front panel internal step.

PHOTO 4: Front panel drivers flush mount.

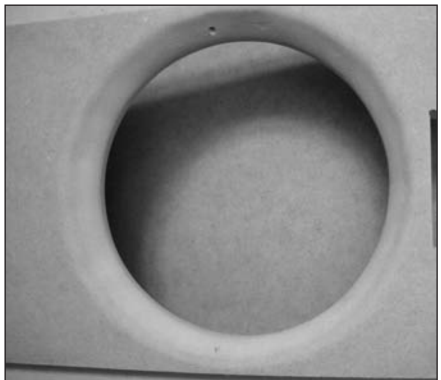


PHOTO 5: Enlarged woofer internal hole.

PHOTO 6: Rubber insulated rivet nuts.

horizontal axis; the fourth mode refers to a panel divided in three sections with the center section moving forward while the other two move backward; and so on.

To reduce the transmission of vibrations from the woofer to the baffle, I used rubber insulated rivet nuts named Well-Nuts or Flex-Loc (**Photo 6**). Actually it's not so easy to find speakers that use these kinds of nuts, even if Vance Dickason[4] has suggested their use for a long time and their cost is not an issue. If you are still not convinced of their utility, I suggest you read Jim Moriyasu's *audioXpress* article[5], in which he proves that using Well-Nuts brought about a reduction of second-third-fourth mode vibrations, while the first mode remained the same.

The chamber's separation panel is 2cm thick, with rails on the lateral panels and embedded in the front-back ones (**Photos 7-8**). The cabinet is brush painted (**Photo 9**), and the dimensions diagram (expressed in mm) is visible in **Fig. 10**.


PHOTO 7: The chamber's separation panel.


PHOTO 8: Front panel internal view.


PHOTO 9: The complete speaker.

TUNING

Auri is tuned at about 40Hz in a net volume of 34.5 ltr, considering both chambers, through three ports (two external and one internal) with 5cm of diameter and 14.5cm of length. The chamber's volume ratio is 2:1, and has been obtained taking into account the drivers and crossover volumes, both located in the bigger chamber V_1.

CROSSOVER

I spent over six months on the crossover design of Auri, alternating computer simulations and listening and measurement, trying to keep the driver phase similar in the driver's crossover region, as well as a dip-free off-axis total response. In **Fig. 11** the starting point is depicted by the unfiltered driver's response, where the Focal 1000Hz and 2800Hz peaks are clearly visible, and the tweeter's 1500Hz peak is followed by a dip. With the woofer I used two parallel RLC cells in series with the signal to take care of the peaks, and a II° order filter. Actually in the second cell, which works on the 1000Hz peak, I didn't use any resistor because the measurement appeared more flat without it. The R2 resistor, which is in series with the C3 cap, is used to vary the filter *Q*.

With the tweeter, after the resistor R3 used to align the driver's sensibility, there is a III° order filter. All drivers are connected in phase with the signal, and their crossover frequency is about 2700Hz. The crossover schematic is depicted in **Fig. 12**.

SPEAKER MEASUREMENTS

Figure 13 shows the system impedance response with the three peaks present in all double-tuned loudspeakers. There are no unwanted resonances, and the load offered to the amplifier is of the easy kind, with the module always more than 6Ω and with an almost positive argument at its minimum. The impedance is a complex number formed by a real part (the module) and an imaginary one (the argument). With some calculations, you can obtain useful information.

Considering only the real part of the impedance and applying the formula $Rs = Z^*\cos(q)$, you can determine the speaker maximum load value. For example, from the impedance chart, frequency (f)= 41Hz, impedance (Z) = 9.9Ω, Phase (q) = +9.5°. Converting the phase degree into

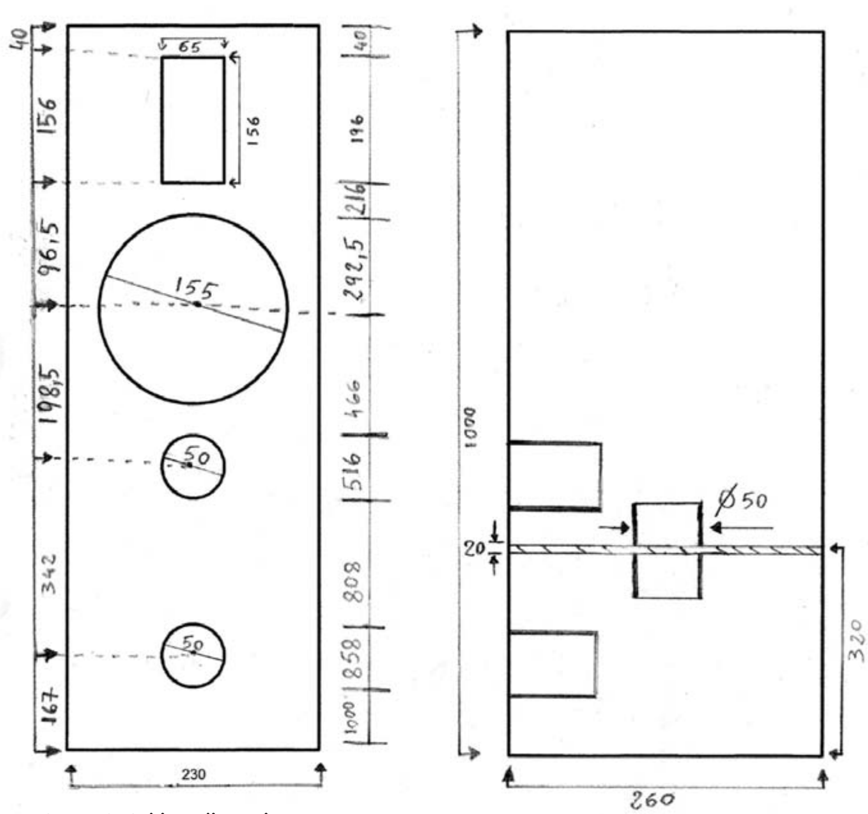


FIGURE 10: Cabinet dimensions.

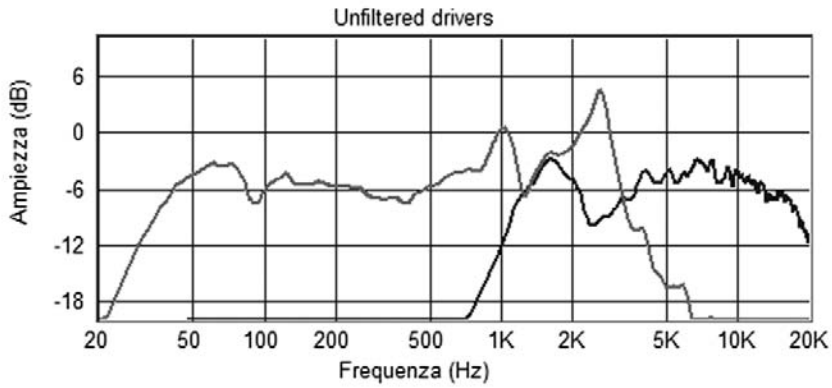


FIGURE 11: The unfiltered driver's response.

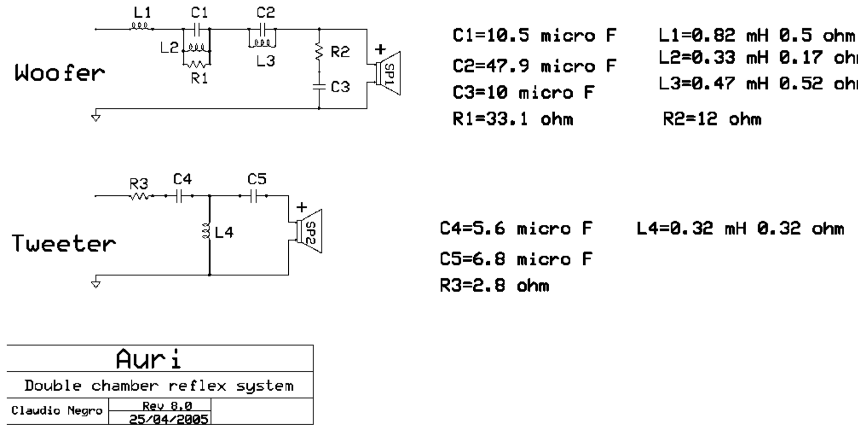


FIGURE 12: The crossover schematic.

radians and applying the formula gives Rs = 9.9*cos(0.16) = 9.7Ω. Thus at 41Hz it's as though the amplifier is connected to a 9.7Ω resistive load.

The Auri maximum load situation happens at 2819Hz with a resistive load of 6.2Ω. You can import your impedance into a spreadsheet, such as Excel and let it do the calculation for you. Moreover, considering both the real and imaginary parts of the impedance, you can calculate the energy requested by the speaker at each frequency by applying the formula *W = cos(q)/(|Z|*2πf)*, allowing you a better vision of what the amplifier will need to give. W = cos(0.16)/(9.9*257.4) = 0.000386 joule at 1V RMS.

I suggest you draw a graph with the calculated active energy values up to 150Hz, because the energy request diminishes as frequency increases, and look at the curve peak in **Fig. 14**. A narrow tall peak means a quick energy request, which is more difficult for the amplifier to handle than a wider curve. Special thanks to Valerio Maglietta, whose help has been instrumental both in the impedance understanding and in the active energy equation.

The one meter semi-reverberant anechoic response is in **Fig. 15**. The first dip is of the DCR itself, while the one at around 1200Hz belongs to the Focal 7K4412.

As a matter of fact, I give more importance to the in-room response, because it better depicts what arrives at our ear. I positioned both speakers in their final room position, along with the mike in the

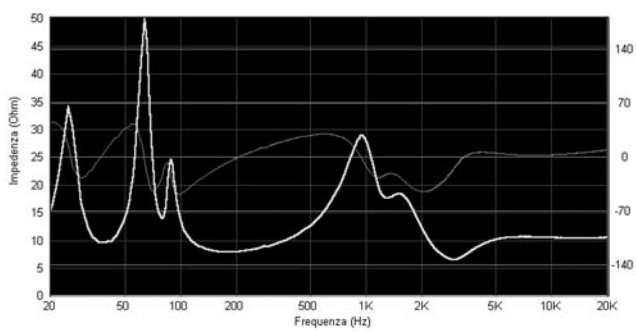


FIGURE 13: Loudspeaker impedance response.

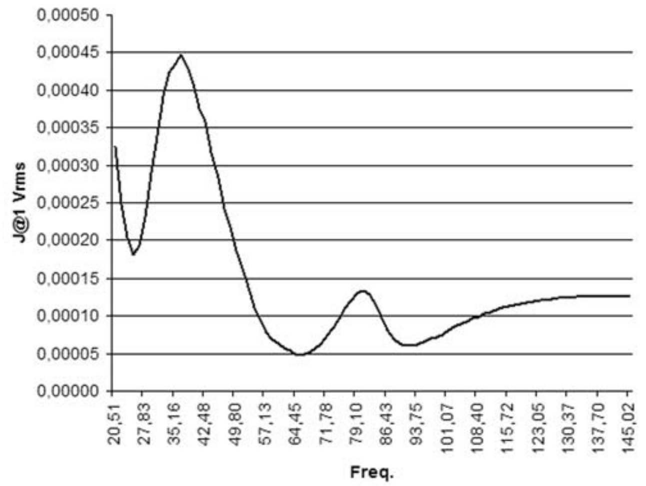


FIGURE 14: Loudspeaker active energy graph.

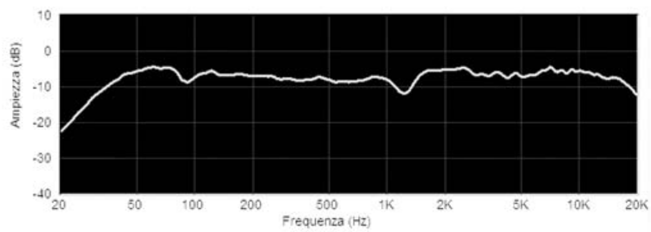


FIGURE 15: Loudspeaker semi-reverberant response.

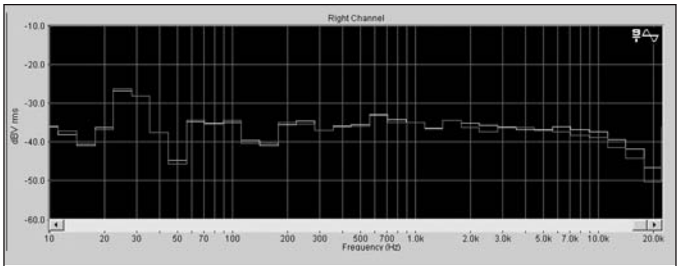


FIGURE 16: Loudspeaker in-room response, on-axis and 30° off-axis.

listening place. Both speakers received a ⅓ octave pink noise signal from two generators, thus each speaker was playing a different signal. In case you don't have two signal generators, you can produce a wave file with the program Sound Forge (which is capable of generating a not cyclic track, different for each channel), and then burn a CD with it to use your CD player as a double pink noise generator. If you are interested, you can download a ready-to-use wave file from my home page[6].

I also find it important to have a reference response for comparison, because it helps understand what is due to the speaker and what to the room acoustic: I used a pair of Dynaudio Contour 1.8 MKIIs for this purpose.

Another big reference help comes from the Henning Møller curve[7], which shows the optimum curve for hi-fi equipment measured in listening rooms. Even though the Møller study was presented in 1974, I think it has been forgotten by most! I have been lucky to know Renato Giussani, the Aedon Audio NPS-1000 designer, journalist for an Italian hi-fi magazine at that time, who personally assisted at the Møller workshop and keeps transmitting this precious information to people, like me, who were too young to remember these AES papers.

Finally, in **Fig. 16**, you can see the in-room ⅓ octave pink noise response, measured on-axis and 30° off-axis with this method.

CONCLUSION

I am satisfied with music coming out of Auri, especially the bass region where the double chamber reflex works very well. I liked the rich sound from the ribbon tweeter, but I will probably go back to using a classic dome for my next project, because I am more used to its sound. I found the best in-room position of the speakers by placing them at least 50cm from the back wall and oriented parallel to each other. Unfortunately, Focal is not selling home components anymore, so it's up to you to search the dealer's stock or to find a valid substitute for the 7K4412. **aX**

Claudio Negro *is a retired commercial airplane pilot, and has been interested in speaker building since the age of 13 when he built a zero offset three-way speaker with a self made tweeter acoustic lens. He also became interested in electronics, building a mixer based on a Radford preamp and a Marantz MM pre-schematics, with separate PSU enclosure. In 1985 he was classified sixth in an Italian magazine DIY speaker contest, after which he took a long break from DIY audio. Recently, he has studied speaker measurements software, publishing a tutorial on Speaker Workshop. His other interests are listening and playing hard rock music, photography, computers, and online car racing.*

REFERENCES

1. http://www.speakerworkshop.com/.
2. http://home20.inet.tele.dk/kou/ubmodel.html.
3. Joseph D'Appolito, *Testing Loudspeakers*, 1998, p. 59; James Moriyasu, "Acoustic Diffraction: Does It Matter?", Feb.'03 *audioXpress*.
4. Vance Dickason, *The Loudspeaker Design Cookbook*, 2000, p. 102.
5. James Moriyasu, "Panel Damping Studies: Reducing Loudspeaker Enclosure Vibrations," Feb. '02 *audioXpress*.
6. www.claudionegro.com.
7. Henning Møller, "Hi-Fi Tests-with 1/3 Octave Pink Weighted Random Noise," AES convention 47 (February 1974), Preprint Number A-5.

Back

Distortion Isolation in the Time Domain

This author explains a novel measurement technique for separating linear and nonlinear effects.

By Bill Waslo

This article describes a technique for separating the nonlinear distortions or noises (harmonic, intermodulation, buzzes and rattles, noises, and so on) from linear effects (such as frequency response, phase response, room reflections) in the reproduced result of any program material. In effect, it allows you to make a recording of a sound system playing a file which has the nonlinearly distorted part separated from the linear, undistorted part.

For those who remember their Fourier theory, the operation of Distortion Isolation is relatively easy to understand. But just in case you've forgotten, I'll start with a simple review.

Remember that if a system is both linear and time-invariant, then its effects on an input signal can be completely characterized by determining the system's impulse response. The impulse response is equivalent to the full complex frequency response, and these two can be converted between each other using the Fourier Transform and the Inverse Fourier Transform. This essentially means that if you know a linear, time-invariant (LTI) system's impulse response (or complex frequency response), then you can, in principle, calculate its exact output signal for any input signal in its passband, as shown in **Fig. 1**.

"Linear" refers to a situation in which a system's effects on an input signal do not change when test levels are changed. You get the same relative change at low levels as at high levels, and the system never distorts, compresses, rattles, or clips no matter how much you scale the input signal. This is an idealized situation, since no real systems or devices are completely linear—even a wire will eventually melt or arc over if driven with a large enough signal. Real-world loudspeakers and amplifiers will produce greater distortions as signal levels are increased beyond some point.

"Time-Invariant" means that the system produces the same output result every time the same input signal is fed to it. The result doesn't change over time, and has no random components. The act of testing once doesn't affect the following tests. Again, this is an idealized situation, as real systems and devices have internal noise that will not be the same each time, and some have memory effects.

In a practical sense, there will also be external noise in any test environment, the effects of which will not be the same each time a test is run. This will be particularly true when acoustical devices (loudspeakers and microphones) are involved.

Because no real audio systems are completely linear and time-invariant, you might think of them as having both an "LTI part" and a "Non-LTI" part, both of which act on any input signal. You could then consider the resulting output signal to be the sum of the LTI and non-LTI parts of the system (see **Fig. 2**).

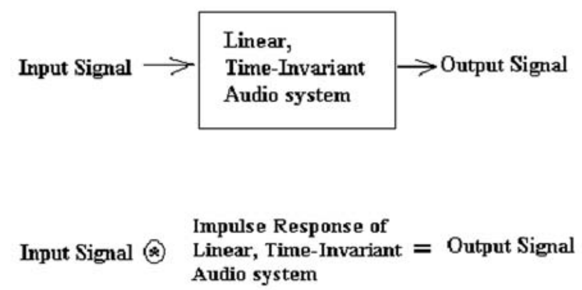


FIGURE 1: LTI diagram.

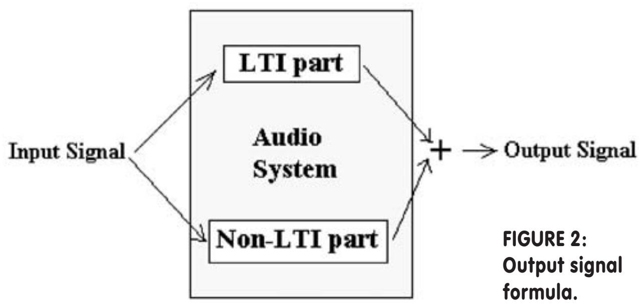


FIGURE 2: Output signal formula.

The LTI part includes all the effects of frequency response, delays, and environmental reflections, which might be mitigated through equalization, room treatments, or directivity control. The non-LTI part includes the harmonic and intermodulation distortions, hums, noises, rattles, and spurious or aliasing tones, which are usually more difficult to deal with. Mathematically speaking, the LTI part is that which can be calculated, given the system's impulse response and a recording of the input signal. The non-LTI part is the portion of the signal that cannot be calculated that way. This, of course, assumes that the system is "approximately linear," and can be said to even have an impulse response when measured at levels low enough to not significantly distort; and at levels high enough to avoid measurement and ambient noise.

If you've been following up to this point, it should be clear where all this is going. If you can measure an impulse response that you can consider as the small-signal characterization of the audio system, then with a given input signal, you can calculate the LTI-only part of the output signal. And, if you record the actual output signal of the system, and subtract the calculated LTI part from it, you are then left with the isolated non-LTI part, as illustrated in **Fig. 3**. The distortion and noise part is isolated, for analysis or audition.

This is somewhat similar in flavor to the Baxandall/Hafler power amplifier distortion tests[1,2], by which you can listen to the difference between the input and a scaled-down output of a power amplifier. An ideal amplifier should reveal no audible difference. Such subtraction techniques can give not only a numerical measure of the difference (the energy in the difference can be determined using some other calculations or test equipment), but they can also provide a way to subjectively experience the difference by itself, to hear the character of non-LTI distorts the system is doing to a given input signal. The new technique described here, however, is also applicable to loudspeakers, listening environments, or entire audio systems as well as to amplifiers.

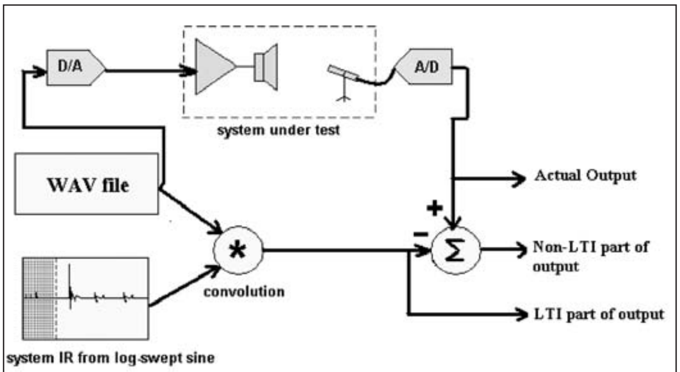


FIGURE 3: Isolating LTI.

REASONS FOR ISOLATING

The distortion isolation process can be useful for a number of purposes. For the equipment or system designer, it can give direction for improvement efforts. If the distortion and noise (non-LTI) part is very low or non-offensive sounding, then significant improvements are more likely to come from equalization, noise reduction, room treatment, or directivity control. If the non-LTI part is high and/or irritating, then effort toward improving linearity or signal level handling is warranted.

Isolating the non-LTI portion will also help in determining the severity or give hints about the cause of problems. Mechanical rattles should be more easily identified. Amplifier clipping has an abrupt, rather immediate sound. Signal compression tends to leave a non-LTI part that sounds much like the LTI part.

Distortion isolation also somewhat mitigates the subjective nature of some listening tests. Humans can still judge how much of a problem the distortion is, but the effects are not as much masked by the linear parts of program material as in a normal listening test. This could perhaps lead to better agreement by listeners about the effects of such distortions, as the effects should be more apparent. Of course, A/D and D/A converters are involved, so there could be some objection about the purity of such results, particularly by those who are wary of digital audio.

This kind of test can also avoid effects of listener fatigue. Sometimes just having the volume too high—even if clean—makes it sound bad. Ears are not perfectly linear, either! Distortion Isolation can help determine whether the problem is with the system or with a subjective listener who is overloaded or just tired of too-high volume.

OPTIMIZING THE RESULTS

You can improve on the test, if you want to emphasize nonlinear distortions but are not as concerned with noises or rattles. Almost any modern synchronous measurement system provides a way to average measurements. This averaging will tend to reinforce the repeatable part of a response, while rejecting the time-variant noises. Averaging is always a good thing when acquiring the impulse response, as it reduces noise in the system characterization.

The acquisition of the Actual Output (the top path in **Fig. 3**) can also be averaged, but doing this will depend on what you are trying to accomplish. You can use averaging to mitigate a test environment that is not free of ambient noise. If you are searching for repeatable nonlinearities, then averaging is good. If you are looking for, say, door panel rattles in a vehicle, then averaging is probably not a good idea, as the rattles may not be the same each time and would then be reduced in the process. In such a case, a quiet test environment is required, as noise reduction via averaging is not an option.

Observant readers will notice that for acoustic systems such as that shown in **Fig. 3**, a measurement microphone is also included. Any noises or distortions due to the microphone will also unavoidably appear in the non-LTI output signal. Unfortunately, for high-level tests that are usually of interest for this process, the only option is an infusion of sufficient money for the microphone.

Typical unmodified electret microphone capsules usually will begin to compress at higher listening levels. For best results, I recommend a high voltage condenser microphone capable of handling high SPL levels. Of course, for non-acoustic tests, this is not required, as only electrical probes are then involved.

For this technique to work well, you need to first acquire a clean impulse response that is indicative of the linear performance of the system being tested under the exact same conditions as will be used for the output acquisition. When testing acoustic systems, the impulse response should be at least as long as the reverberation time of the system and room. The impulse response should also be measured at relatively low input levels, to avoid compression of the output signal. The best method to use for acquiring the impulse response is the "Log-Swept Sine" technique, which has the ability to eliminate harmonic distortion in the measurement[3], leaving a very pure impulse response result.

FIGURE 4: WAV player in PRAXIS.

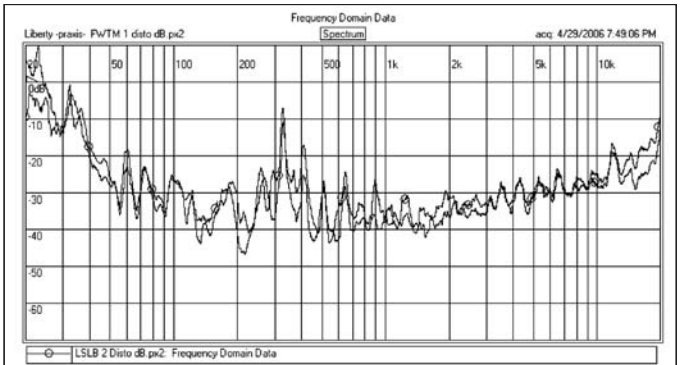


FIGURE 5: Distortion plots made with program material.

THE PROCESS

The distortion isolation (time domain) process has been largely automated for the Liberty Instruments "PRAXIS" system, in its "Distortion Iso (Time Domain)" script. The script guides the user and runs through the process as follows, performing the setups and acquisitions.

First, you must select the program material to be used. It should not be too long, to minimize the time required to later calculate the expected LTI portion of the output signal. WAV files of half a minute or less are manageable with most computers. The file is played through the system in a dry run, so you can turn up the volume to the desired levels (usually to where you think you hear some objectionable effects) and can adjust the input gain levels of the measurement system (in the "Levels Form") to optimize its dynamic range. These settings must then remain stable through the rest of the process.

Next, you make an impulse response measurement, preferably using some user-configurable averaging, via a Log-Swept Sine ("Log Chirp") at a reduced drive level. This impulse response should be made immediately before the acquisition of the actual output of the system, to reduce the possibility of changes in air pressure, microphone position, or electronic component drift causing error in the calculated LTI portion. It is important that the measured impulse response is of the system output with respect to the internal digital signal path of the computer measurement system, including all gains and delays, measured as exactly as possible. The A/D converters and D/A converters of the measurement system, like the microphone, are all part of the system being measured, and must be included in the IR measurement. In PRAXIS, a special "Chirp (synchronous)" measurement stimulus is provided expressly for this purpose.

The software then combines the WAV file and the measured impulse response data using an operation known as linear convolution. The result of the convolution is a sound recording that represents what the A/D output would be if the program file were played through an LTI version of the system that is being measured.

Finally, the WAV file program material actually is played through the system, and the result is digitized and recorded by the measurement system. If only time-invariant phenomena is being investigated (that is, if noise or rattles should be reduced), then this recording can be repeated and averaged as many times as you wish. After this, the expected signal is simply subtracted, sample by sample, from the measured signal, leaving the non-LTI signal.

OUTPUT FORMAT

The PRAXIS script provides this in a format in which the left channel (channel 1) contains the LTI portion and the right channel (channel 2) contains the non-LTI portion. Summing the two, of course, gives the entire output signal as it was recorded. You can save the data in PRAXIS' "px2" format, or export it as a stereo WAV file.

The result files are best auditioned using headphones. They can be played using various media players, or you can use the PRAXIS program (in its full or free/demo operation) for this. You can download PRAXIS from www.libinst.com/praxis_downloads.htm at no charge, and use it to examine, further process, or listen to example files produced using the distortion isolation method.

To use PRAXIS to listen, load the WAV file(s) into the Primary Plot using the menu "File, Open" and selecting "Files of Type" = "wav" and browsing to where the files are stored. After loading the file (it may take a few seconds, since the files are large), select the menu "File, Listen" on the Primary Plot form, and select the soundcard to use. You can open the soundcard's mixer to select inputs or adjust volume. With stereo WAV files, there will be a series of buttons on the player labeled "m," "s," "L," and "R."

Before starting the playback, select "m" to hear both channels combined in mono (for a distortion isolation file, this will be the same as the actual output signal). Select "L" to hear the LTI portion. Most interestingly, select "R" to hear the distortion and noise (non-LTI) portion by itself. Click on the blue arrow at the left of the form to begin playback.

SOME EXAMPLE FILES

I had a request from a potential user in the automotive industry to generate some distortion isolation files with a car stereo system in a vehicle. The following files were generated using PRAXIS 1.37c, with Audpod and an M-Audio Transit card and an ACO Pacific 7012 microphone (with the "PS9200 Kit"). The vehicle was a 2005 Chrysler Town and Country van, with a six-speaker sound system. The deck in the van was a JVC KD-SX980 (50W × 4). This vehicle was used because it had a line-level input on its panel. The computer was an older 650MHz Compaq laptop.

I used two WAV files. The first was about 30 seconds of the cut "Lie Still Little Bottle" by They Might Be Giants from the *Lincoln* CD. The second was from the cut "First We Take Manhattan" from Jennifer Warnes' *Famous Blue Raincoat* CD. I selected these because of their strong bass lines and reasonably clean sound.

We tried to get the cuts to stimulate some strong panel vibrations in the van. We could make the speakers sound overdriven, but buzzes were not apparent. The vehicle door panels were (disappointingly, for our tests) well behaved. During the second take on the Jennifer Warnes cut, we also leaned a tray of silverware against a door speaker and tested without averaging to try to bring out some buzzes, with slight success.

In most of these cuts, a hum is audible in the background. This is from operating lawn mowers in the neighborhood, the tests being done in a suburban setting on a Saturday afternoon. The program material was fed in mono, simultaneously to all loudspeakers in the van.

You can download and listen to the files listed below via the Internet link

FILE	DESCRIPTION
LSLB original program.wav	"Lie Still Little Bottle," single channel, no processing
LSLB DI Low Level.wav	"Lie Still Little Bottle" distortion isolation file, at moderate listening level, 4× averaging
LSLB DI High Level.wav	"Lie Still Little Bottle" distortion isolation file, at about 6dB higher level. Some clipping evident on singer's voice. 4× averaging
FWTM original program.wav	"First We Take Manhattan," single channel, unprocessed
FWTM DI High Level.wav	"First We Take Manhattan" distortion isolation file, at high level, 4× averaging. Speaker overload evident, slight buzzing.
FWTM DI Mod Level.wav	"First We Take Manhattan" distortion isolation file, about 3dB lower, no averaging on actual recording. With silverware tray added to try for extra buzzing!

www.libinst.com/distortion_isolation_in_the_time.htm. You can listen through PRAXIS, or nearly any media player. Remember that the left channel is "LTI," the right channel is "non-LTI" (distortion and noise). Also, all files have been "normalized"—that is, scaled in level to best use the 16-bit data format, so don't expect the "6dB higher level" to be 6dB higher when you listen.

ANALYSIS AND COMMENTS

One oddity I noticed was that there is a point early in the "LSLB DI" recordings in which it sounds to me as though there is a breakup or distortion. The same sound is not evident (to me) in the original program file. I would have thought this was definitely nonlinear distortion, but it appears in the LTI (left) channel! Apparently my ears aren't as good at differentiating linear frequency-response effects from nonlinear distortion effects.

The addition of the silverware tray didn't seem to make too much difference in the last test. It was not noticeably rattling inside the vehicle, "live," either, but is mentioned in this report for completeness.

Files from these kinds of tests can be further processed with PRAXIS to provide more objective results. For each of the two "DI High Level" files, I separated the right and left channels, FFT'd them (to find their spectrum), and then smoothed the results with 1/12th octave smoothing. Then I divided the right (non-LTI) spectrum by the left (LTI) spectrum of each file to get a plot of relative "dB distortion" using program materials as stimulus. The graphs showed lower values than I would have expected, with noise and distortion being about 35 to 40dB down through most of the midrange.

At lower frequencies, the curves rise significantly, probably because those frequencies are far below the cutoff of the loudspeakers. At higher frequencies, the curve also rises, which could be due to more distortion, or because of greater noise. This latter rise may also be because harmonics of individual tones in the program material will push toward higher frequencies.

The most surprising aspect of these two plots is their similarities. Both spectrum curves are overlayed in the PRAXIS plot (**Fig. 5**). Considering that they were made using different program material files, the similarity is remarkable. *aX*

Bill Waslo *heads Liberty Instruments (libinst.com), developer of the PRAXIS audio measurement system.*

ACKNOWLEDGMENTS

Thanks to Joachim Gerhard for his ideas, efforts, and bug reports about the Distortion Isolation script in PRAXIS. Thanks to Tom Alverson for his assistance in setting up and making the example files, and for the use of his van.

REFERENCES

1. Baxandall, P., "Audible amplifier distortion is not a mystery," *Wireless World*, November 1977, pp. 63-66.
2. Hafler, D., "A Listening Test for Amplifier Distortion," *HiFi News & Record Review*, November 1986, pp. 25-29.
3. Farina, A., "Simultaneous Measurement of Impulse Response and Distortion with a Swept-Sine Technique," presented at the AES 108th Convention, Paris, France, 2002 February 19-22.

DVD Is Audiophile Grade. . .

But Check The Fine Print

By Jesse W. Knight

My knee-jerk reaction to anything video is that the sound must be bad. KILL YOUR TV is one of my favorite bumper stickers.

When a TV in our house died, I replaced it with one that has a built-in DVD player and that all-important headphone jack, knowing that our library system was adding DVDs of plays, movies, concerts, and opera to its collection. With its 22,000 DVDs and counting, I now have a no-cost way to check out DVD quality. Library loans are for one week, which is long enough to decide whether or not to add an item to my collection. Here's what I've found.

In a nutshell, if you can test DVDs and mid- to high-end players before you buy, and are willing to experiment with speakers, fantastic sound can be yours. After obtaining a basic player, explore the possibilities of inter-library loans at your library. DVD rental is the next best option, and you can find specialty DVD rentals online. When all else fails, there are always reviews, although most reviewers are not critical audiophiles.

In my opinion, opera presents a worst-case scenario for audio gear. Other music may not exhibit as much distortion as a consequence. Everything except the DVD itself is troublesome. You might not agree with my priorities for assigning the degree of trouble each topic involves, but here is my opinion.

1. **Speakers have always been trouble.** With the need to correlate stereo to the screen, they just became more troublesome. Most complaints about DVD sound posted on the web are due to poor speakers. Web postings complaining about excessive dynamics in opera DVD are excellent examples.
2. **Poor DVD player digital decoding.** DVDs have multiple audio playback options; LPCM is 48kHz sampling with 16 linear bits. Then come the perceptually encoded formats—Dolby digital 2.0, Dolby digital 5.1, and DTS 5.1. Finding a player that is clean on all DVDs in all formats may be difficult. I have found plenty of player issues and no clear-cut disc problems. Few listeners have heard these problems, which I find quite audible. This is not something elusive such as jitter or lower bit errors; these are major design flaws at work.

I always have my headphones and a few demanding DVDs with me when I evaluate players in the field. My low-end player "flags" DVDs for further testing at stores. So far the DVDs receive much better grades than the players do. Finally, perceptual encoding may not be totally imperceptible, even if it cannot be heard.

3. **Microphone placement in complex live performance recordings.** There are unlimited problems when stage directors go off the beaten path. Singers running, crawling, rolling, and popping in and out of trap doors in the stage are hard to track with spot mikes.
4. **Many DVDs made from TV videotapes were dynamically compressed**. Not all that is old is of poor quality; some video productions as early as 1980 took care to preserve most of the dynamic range of the performance and full frequency response. This is a problem that gets very little discussion on the web. Many people notice over-compression of video.
5. **Bad performances.** This is a thorny subjective topic that suggests that your library is your best friend.

ANALOG SOUND ON FILM OR VIDEO

Before DVD, having a picture and audiophile sound at the same time was not practical. I did enough experiments with 16mm projectors in the early 70s to realize that at 24 frames a second neither an optical nor magnetic soundtrack could be very good. I totally redesigned the optical pickup and amplification in a 16mm projector and achieved very disappointing results.

At work I rebuilt several magnetic sound 16mm projectors, finding that there just was not room for a decent sound track. If you got flat response to 7kHz, you were doing well. Wow and flutter were out of control.

Half-helical scan open reel videotape did not improve things much, as the linear tape speed was slow. Sony hi-fi beta had possibilities, but the format did not catch on for home use. The bottom line was that if you wanted sound that was good enough for classical concerts or opera, the picture had to go. Sound on VHS, in my opinion, was marginal only for rock concerts. Well-made rock recordings on vinyl, such as Pink Floyd's *Dark Side of the Moon*, could not be equaled on VHS.

Meanwhile, TV networks worldwide were archiving professional-grade videotape and 35mm films of concerts and opera. Unfortunately, there was little incentive to record sound with a full dynamic range, because TV broadcasting required compressed sound. Likewise, it was probably believed that frequency response beyond what the multicellular horn on the typical theater speaker could reproduce was unnecessary. If a soundtrack sounded good on an Altec A7 "Voice Of The Theater" speaker, it was a "good" soundtrack. This speaker, with its stiff woofer and "squawking" midrange without a tweeter, is not something I would choose to listen to for more than two minutes.

DIGITAL RECORDING

Even the most primitive digital recording unit, the Sony F1, provided a time-stable way to record much better sound with a picture. Some concert videos from the 1980s are very good, when their producers gave some thought to recording for posterity, rather than just what was required for TV. Conductor Herbert Von Karajan set up his own company, Telemondial, with the future in mind. His video of the Brahms German *Requiem* from 1985 has very good sound that is equal in overall quality to sound-only recordings of the same period of this work.

Although the Brahms *Requiem* is musically very complex and demanding of performers and engineers, the singers stand in one place and always sing facing forward. Opera singers move and sing at the same time, which complicates the microphone and mixing demands. In my sampling of DVDs there is some correlation between poor sound and novel staging.

TWO DVD SAMPLES

"Great Conductors of the Third Reich, Art in the Service of Evil," from the Bel Canto Society (BCS –D0052), clearly states on the front cover that the contents are "A compilation (not a documentary), 1933-1943." The back cover clearly explains that this DVD is taken from Nazi film footage. Packed with this controversial DVD is a statement that perceptually encoded DVD formats are not adequate when restoring historic recordings (**Photo 1**).

Here is a clear statement that even historic material deserves uncompressed LPCM recording mode. It is quite possible in my opinion that perceptual encoding, when used on distorted noisy material, does not perform well. The DVD gets high marks for explaining what it is (on the outside) and why the sound was not perceptually encoded on the inside. This DVD is valuable for any study of musical ethics when combined with other resources.

In LPCM mode Verdi's *Otello* (DGG 073 092-9) is a clean highly dynamic recording with excellent picture. Even better, the performance is superb (**Photo 2**). Some web postings have complained about excessive dynamics, but these are from people who tried to listen to this DVD on built-in TV speakers showing that they have no knowledge of audio.

Two important details are buried in fine print inside the package. First is the actual recording date, stated as production date of Oct. 1995. Second is the fact that the 5.1 sound options were derived from a stereo master by using "Ambient Surround Imaging" developed by Emil Berliner Studios.

I don't find any malice in this; rather

PHOTO 1: The controversial Third Reich music video.

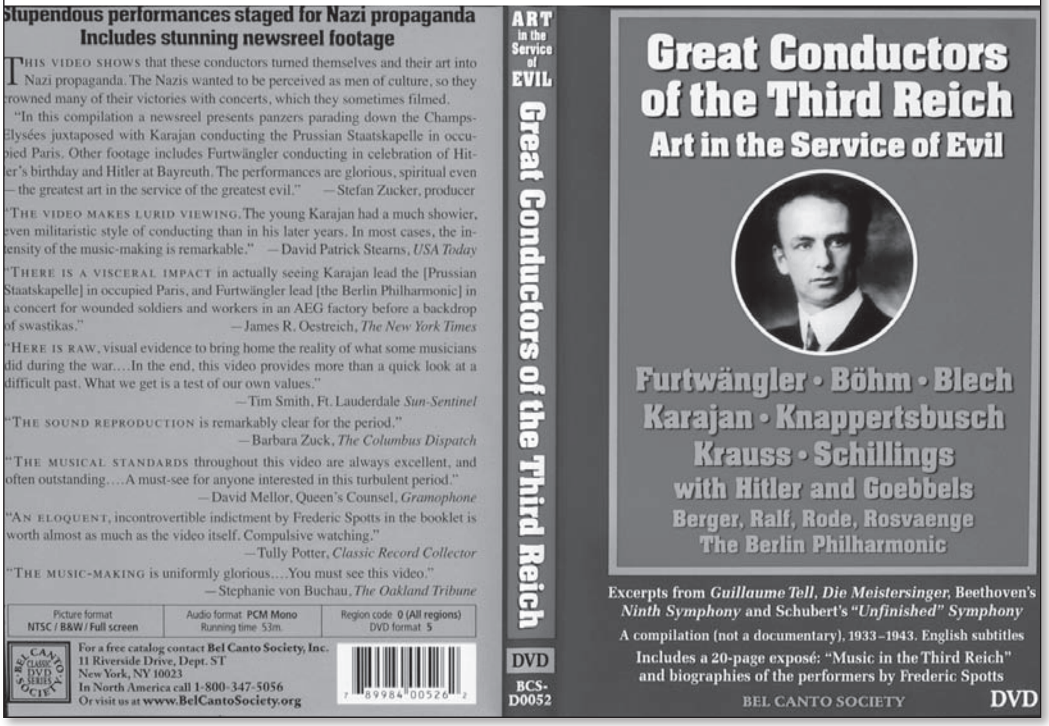


PHOTO 2: Verdi's Otello.

it is an unintended statement on the relative unimportance of sound engineering in a video age. I want to know what I am listening to, without digging through the fine print. Could we have a symbol for 5.1 that is derived from stereo? There is plenty of newer stuff that has far less sonic quality, so I would not pass this over based on age or lack of real 5.1. Highly Recommended.

WIRELESS VS. FAR-FIELD MICROPHONES

Popular music singers use wireless or handheld microphones on stage, and as a result they can move around without changing the sound mix. In a sense, they are electronic instruments that can be panned to any sonic location. Reverb can be added to give a sense of distance. This is a very different vocal technique from what a classical singer learns. I have failed to find a way to audition a popular singer's voice "unplugged" from the perspective of a wireless or handheld microphone. How can I put both my ears at the same time 1″ from a singer's mouth?

Distortion issues with near-field vocal microphones are monumental compared to any DVD audio artifacts from jitter or sampling rates. This is not to say that I think this is bad singing, but it is singing that has no live reference in my opinion. Popular music listeners can still make judgments about DVD players using two identical DVDs and an ABX tester.

Classical voices do not sound good when the microphones are almost in their mouths. Often classical singers are positioned 5-10′ from any spot microphones. When these spot microphones are turned up too high, it can rob the orchestra of power by favoring the voice too much.

If the singer is running around, crawling on the floor, turning his/her head, or singing into another singer's body, the sound perspective keeps changing. Even in the 1990s the sound engineers could not keep up with these antics in productions that had open trap doors in the stage with singers singing waist deep in the stage. Despite this, an experienced listener can make judgments about DVD sound when singers are in their normal upright positions.

LASERDISC

Laserdisc should have ended the practice of overly compressing the audio at the front end of video productions. Unlike London-Decca, TV production crews still did not see the value in making an uncompressed audio recording for future use. Back in 1955, Decca was making essentially uncompressed stereo opera recordings, which, remastered on CD, are quite good, despite crude stereo imaging and some "shimmer." Put another way, they are quite enjoyable.

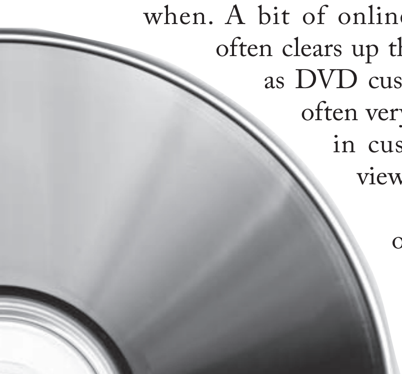
A NEW DVD IS NOT ALWAYS NEW

DVD manufacturers have been searching through the archives of TV stations and film studios for old material that can be issued at less cost to them. Some of this material has historic value that is beyond question. Furtwängler conducting Mozart's opera *Don Giovanni* at Salzburg in 1954 is a good example. Restoration improved the sound, but it is still way below what London-Decca was doing at that time.

For instance, you can compare Lisa Della Casa singing on this DVD to a "Decca Legends 1953" CD of Lisa Della Casa singing Strauss songs and opera excerpts. Nobody would prefer the DVD to the CD based on sound alone.

DVDs have several copyright dates that apply to DVD mastering, restoration, translations, subtitles, and staging. Buried in the fine print, you may find a date for the original sound recording, which may be much older than the video, if the production was lip-synced to an existing recording. Unless your eyes are in fantastic shape, take a strong pair of glasses to the store, if you intend to figure out what was done when. A bit of online research often clears up these issues, as DVD customers are often very articulate in customer reviews.

In my opinion, the best tests



of DVD sound and players for most listeners are the large-scale oratorios of Bach, Beethoven, Brahms, and Verdi. Mahler's symphony #2 is scored for a huge orchestra, pipe organ, and chorus with two soloists. No music has more dynamic range than Mahler's.

VIDEO ISSUES

There is a trend in theater and opera toward an ugly production style in which black paint and fabric predominates. It is cheap and ugly and certainly provokes divisive opinions from audiences and critics alike. It is often fascinating and not totally worthless by any means. DVD is a relatively inexpensive way to explore this phenomenon, but there is a catch.

Eurotrash often violates every rule of using light, whether you are a painter, film photographer, or digital video maker. There is a limit to how much contrast or dynamic range you can capture. A scene with excessive black areas and blinding highlights defies recording.

DVD cannot record what cameras cannot image. Whatever impression a scene with excess contrast makes in the theater will be lost in the camera. This is not the fault of DVD. It is already evident that Eurotrash opera productions are not popular with American home audiences, judging by customer reviews and sales figures.

Rameau opera was not a big seller until William Christie staged a spectacular production in Paris of *Les Indes Gallantes* (Opus Arte, OA 0923 D). This is a ballet and opera combined, lit and staged in a way that the camera loves. There is constant movement and dance, with eye candy galore. This is state-of-the-art in every way for 2003, including the sound.

CONCLUSION

DVD is far superior than any other format in bringing a wide range of theater into our homes. In every case I believed that the limitations of source material were greater than any DVD limitations.

I am concerned about the widespread use of perceptual encoding. Can we be sure that the missing sound does not register in our brains at any level? Will constant exposure to MP3 and Dolby digital formats over time change the way we perceive sound?

As long as an LPCM option is included in a significant number of DVDs, and CDs are still available, we can continue to have all the masked sounds whether we perceive them or not. What is certain is that we can no longer have simple tests for audio quality, due to format diversity. *aX*

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This project would not have been possible without the vast DVD collection at the Minuteman Library Network. Andrew J Nittoli, at the State University of New York, shared insights on perceptual encoding research he is doing to determine whether this encoding is truly imperceptible to our brains. Radio Shack in Woburn and Tweeter in Burlington provided considerable help in evaluating home theater components from Onkyo, Rotel, and Cinevision. These mid price components provide a substantial improvement over built-in DVD players.

Jesse W. Knight has designed custom analog audio gear for Select Film Library in New York. He built a record cutter of his design and was involved in opera and choral recording work. He experimented with a tape/slide opera recording format, designed and built specialized optical systems for 35mm photography, and built several speaker systems for classical listening. He studied classical singing and has been a bass section leader for amateur choirs. He is currently researching the impact of DVD on opera from every possible perspective, as an audio engineer, photographer, and singer.

Back

How To Judge Speaker Quality With Listening Tests

By Larry Klein

Many years ago, when I was young, innocent, and unmarried, I decided that any well-rounded bachelor should have at least a passing acquaintance with the fine art of gourmet cookery. A great believer in the printed word, I immediately betook myself to a local bookshop, where I learned that an important part of almost every serious cookbook is devoted to the purchase of meat. Invariably, after a short essay on what part of which animal produces what cuts, each cookbook made the point that a good beef steak—of prime interest to us all—is "well marbled" with fat. Knowing that no novice in the steak-purchasing art would have in his mind an accurate mental image of a well-marbled steak, the editors of the cookbooks were usually kind enough to provide a badly reproduced color photograph of an adequately fat-marbled cut of meat. That was many years ago, but to this day I have yet to see in any butcher's showcase a steak that resembles in whole or in part any of those illustrations.

What has all this to do with speaker shopping? The relationship is closer than you might think: how many times have you been advised in the pages of this and other audio publications to listen for a speaker that has a "non-peaky mid-range," a "non-boomy bass," a "good transient response," and so on? None of these terms, I'm afraid, adequately convey the particular subjective-objective phenomena that the author is seeking to describe. For unless the reader already has in his mind's ear the sound of a "smooth, uncolored" speaker, he is in no better position to select a pair off the showroom shelf than I am to select a well-marbled steak from a butcher's showcase. This has nothing to do with the much-abused subject of psychoacoustics, but simply relates to the very real question of how one can judge whether a speaker is doing a good job of reproduction when one does not have available for comparison the original sound the speaker is attempting to reproduce.

There is a popular (but fallacious) notion that it is enormously helpful, as a preliminary to speaker shopping, to attend a concert of live music. Then, with the live sound still reverberating within your cerebral cortex, a quick after-trip to a hi-fi showroom will permit you to make instant and accurate distinctions between "real" and "unreal" speakers. Unfortunately, the average mind's ear cannot be expected to isolate the distinguishing features of live sound under concert-hall conditions in the first place. And in the second place, the mind's ear is a notoriously leaky vessel and cannot retain the memory of what it has heard for more than a few seconds even under the best of circumstances.

Some experts maintain that A-B listening comparisons of equipment are of limited value: they are rather like attempting to evaluate the accuracy of two reproductions of a painting, each differing from the other in color values, without having the original painting at hand. This view has merit, I feel, only when you are dealing with very subtle differences among several very good loudspeakers or when you are attempting to educate an untrained ear. These exceptions aside, I have not found the task of speaker evaluation by listening at all difficult with the usual run of speakers that find their way into my living room.

The average speaker buyer, of course, will seldom have the opportunity of hearing a large number of models in his own listening room. He almost certainly does not have an A-B-C-D switching setup built into his hi-fi system. Nor is he likely to have made up a test tape of program material chosen specifically for speaker evaluation. Nonetheless, it is possible to make a valid judgment of the sound quality of a speaker in a home or showroom once you become aware of what to listen for and how to listen for it.

Most speaker manufacturers seem to have pretty well licked the bass- and treble-reproduction problems: fewer systems are now being produced with woofers whose bass is lacking or overly boomy, and similarly, fewer systems have tweeters whose treble output is inadequate or screechy. Both bass and treble performance can be evaluated in the showroom with a few simple checks.

In my own listening tests I find it easier to make discriminations if I'm listening to one speaker rather than a stereo pair. However, one runs the slight risk, when auditioning a single speaker in a showroom, of having its location influence its frequency response. This can be easily checked by switching to the other member of the pair, which probably will not be installed in as advantageous (or disadvantageous) a location.

An important aspect of a speaker system's bass performance is its freedom from spurious resonances. This can be tested simply by tuning in several FM stations and listening carefully to the various announcers on the speaker(s) under consideration. One or two of the announcers may have naturally deep voices, but if every one of them sounds as if he were addressing you from the bottom of an oil drum, you can be sure that the loudspeaker under test (not the announcer) has a bass resonance peaked somewhere in the 100Hz region. This resonance provides for some pleasant overlay of bass on classical material and enhances the beat on pop stuff, but the price paid for this is loss of upper-bass clarity and (usually) absence of genuine low bass.

A speaker system that is inherently unable to reproduce low bass has a choice of how to react: when fed, say, 40Hz, it may either not make any sound at all (which is considered preferable), or produce sound that has within it a small amount of the original 40Hz signal, some 80Hz, and a lot of 120Hz noise. The spurious 80 and 120Hz harmonic signals are products of a phenomenon known as doubling or tripling that occurs when a speaker cone exceeds the linear limits of its suspension. With such a speaker, the sonic difference between a kettle drum and a trap drum, for example, is completely obscured.

As far as the high-frequency performance of the speaker system is concerned, a good test is to listen to recordings of music that include cymbals or triangles and try to isolate the ringing or shimmering sound that is typical of those instruments. You will probably have to listen carefully for this quality in several speakers before you can easily distinguish between those that have it and those that don't. Note also, while listening for shimmer, the amount of FM or tape hiss present. You'll find that some speakers will have the shimmer plus a liberal helping of hiss, and that some will have the shimmer without the noise. Emphasized hiss is usually the result of an irregularity in the speaker's high-frequency response, which may or may not audibly affect other aspects of its performance.

Another quality essential to good tweeter performance is wide dispersion—its ability to spread the high frequencies in a broad arc across your listening room. Without it, the sound will be closed in or boxy; with it, there is a sense of openness and airiness.

You can check for adequate high-frequency dispersion by using the interstation noise on an FM tuner. Switch off the interstation muting if present, and find a weak station or an unoccupied section of the band. Listen for the interstation noise (it has a rushing, hissy quality), while standing directly in front of the speaker. Then, concentrating on the hissy quality in the sound, walk off to one side of the speaker system; at some point you will find that the hissy quality disappears. You will find the same effect on the other side of the speaker—and probably, depending on where the speaker is installed, the hiss will also diminish if you duck your head toward the floor.

The wider the area covered by the very high-frequency hiss, the better the high-frequency dispersion of your speaker system, and the more open and natural-sounding will be the music reproduced by it. In addition, the problem of speaker placement, speaker separation, and listener position will be far less critical than with a narrow-dispersion system. This test should only be made, however, after you have already established that the speaker's on-axis high-frequency response is everything it should be—if it is already short on highs, you may not notice any falling off at the dispersion limits.

There is one area in which many speaker manufacturers still run into bad trouble—the mid-range. Good reproduction of the middle frequencies does not necessarily require a separate mid-range speaker.

In the past, a number of manufacturers (thankfully there are fewer each year) deliberately set out to design speaker systems with a 5 to 15dB boost in the upper middle frequencies. These manufacturers were—and are—appealing to the audio equivalent of the thirteen-year-old mentality oft cited as the target of most TV fare. On certain types of material the mid-range boost imparts a sense of projection, a front-row-center quality, to everything played. This is the so-called "presence" phenomenon, and it is a hindrance, not an aid, to accurate reproduction.

The overly bright and projected quality that comes from a boosted mid-range may be impressive on first listening, but it is accompanied by unfortunate side effects. Minor side effects are harshness (depending upon the range of the frequencies boosted) and emphasis of high-frequency noise and distortion in the program material. A major side effect (for me, it can make a speaker unlistenable) is a kind of nasality or honkiness that accompanies and discolors everything the speaker is reproducing. Not to be accused of slipping back into well-marbled meaninglessness with the terms "nasality" and "honkiness," I have worked out a "live vs. reproduced" technique that makes it possible for anyone to imitate—and to detect—this type of objectionable coloration.

First, set up an FM tuner as you did for the high-frequency dispersion test. Then cup your hands over your mouth (as though you were trying to warm them with your breath) and make a "shhh" sound. Now remove your hands and make the same sound. Repeat several times until you hear the difference. The hollow, rather nasal quality heard with your hands in front of your mouth is a good approximation of the nasal or honky quality associated with midrange difficulties in a speaker.

If you are in a hi-fi showroom, have the salesman switch among a number of speakers while you are listening to interstation noise. You will soon be able to pick out the speakers with the nasal quality rather quickly. And with practice you should be able to detect this same quality on music and voice if the speaker has a particularly bad case of the honks.

It is well to remember that if a speaker is suffering either from bass boom or a honky mid-range, the effects will pervade *everything* coming through the speaker. For this reason, it is relatively easy to determine whether the fault lies with the speaker or with the program material.

There are numerous other aspects of speaker performance and misperformance that an experienced ear can evaluate, but their subtlety precludes accurate verbal description. For example, I have heard speakers that I can only describe as having a "gritty" quality, a kind of harshness in the treble range—but the word "gritty" does not quite describe it. The only way I can pinpoint this quality for another pair of ears is to put on a recording that I know will demonstrate it and say, "There, listen to what is happening in the upper register of the piano."

The crux of the difficulty in evaluating speakers is *knowing what to listen for*. I have had this fact demonstrated to me a number of times during meetings of the New York Audio Society. I once, for example, switched from one to another of four under-$100 speaker systems under test while playing a Frank Sinatra recording. I asked for a vote as to preference, and the opinion of the 15 people present was about equally divided among the four systems.

I played the record again and asked the listeners to note that three of the systems, to a greater or lesser degree, made Sinatra sound as though he were singing through his nose, while one of them did not. Immediately, everyone was able to hear the superiority of one of the speakers (the non-nasal one) over the other three. (One can safely assume that the voice of Frank Sinatra does *not* suffer from nasality.)

Power of suggestion? No. And it wasn't a matter of the audience's being unable to hear the difference; they simply did not know what to listen *for*. And that, in a nutshell, is the problem I hope this article has helped to solve. **aX**

From a subsequent issue of *HiFi/Stereo Review*:

Speakers

• Larry Klein's "How to Listen to Speakers" in the August issue is a gem. I think it will be most helpful to a great many of your readers. At the same time, it can do nothing but help manufacturers of good speakers.

Roy F. Allison, *Plant Manager*
Acoustic Research
Cambridge, Mass.

Back

Jensen Transformers
9304 Deering Avenue
Chatsworth, CA 91311
(818) 374-5857
www.jensen-transformers.com
Dimensions: 4.32″ W × 2.75″ D × 1.45″H
Net weight: 11 oz

Inside the ISO Max

By Charles Hansen
and Gary Galo (listening tests)

This article revisits the Jensen CI-2RR Dual Audio Ground Isolator that was used to solve a few audio problems in the 6/00 issue of *Audio Electronics* magazine[1]. This time around, Gary Galo will do a more thorough audition, and Charles Hansen is going to run the suite of test measurements.

The CI-2RR is a stereo ground isolator designed to stop hum and buzz in unbalanced audio interfaces by eliminating the inherent ground noise coupling mechanism. **Photo 1** shows the front panel, which has two pair of high-quality gold-plated RCA phono jacks. The CI-2RR is constructed of heavy gauge painted steel.

PHOTO 1: CI-2RR front view.

Photo 2 shows the CI-2RR with the cover removed. The input and output jacks are isolated by two Jensen JT-11P-1HPC isolation transformers with nickel-iron-molybdenum alloy cores. A Faraday shield between the windings connects to the steel transformer case and provides maximum RF interference rejection.

PHOTO 2: CI-2RR interior.

A schematic of the CI-2RR is shown in **Fig 1**. The secondary winding has a series RC compensation circuit placed across it.

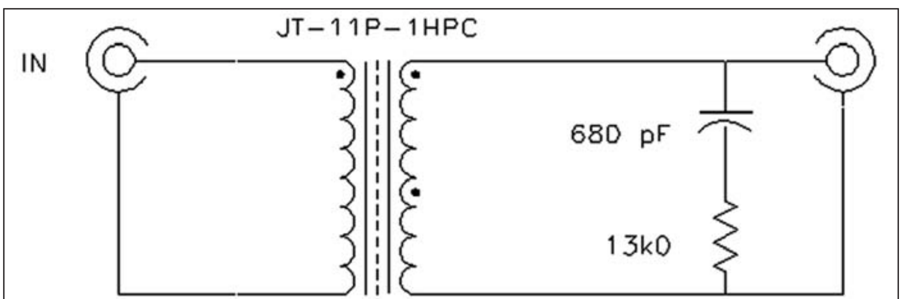

FIGURE 1: CI-2RR schematic.

MEASUREMENTS

The CI-2RR maintains normal polarity. The input impedance measured approximately 48k at 1kHz, with a DC winding resistance of 2.18k. The output impedances for both channels measured approximately 4k2 at 1kHz, with a DC winding resistance of 1.90k. Winding-to-winding capacitance was a low 121pF and winding inductance was over 800H, which should enhance low-frequency response. The leakage inductance (secondary shorted) was less than 1µH, and the capacitance between the two inputs measured a low 15pF, which should minimize the crosstalk between channels.

I recorded the output frequency response (**Fig. 2**) for resistive loads of 100k and 47k with an input source impedance of 600Ω as specified in the datasheet. At the nominal specified load of 47k, the frequency response for the CI-2RR was within ±1dBu to 26kHz, with 0dB defined as +4dBu (1.228V RMS) at 1kHz into 47k. I also took data for loads of 10k, an IHF line load, and 600Ω; but even at 10k load the response was -7.6dB below the 47k load response curve at 1kHz and is off the graph.

Insertion loss at 1kHz with a 47k load was 0.80dBu, and crosstalk performance was –59dBu at 20kHz. The common-mode rejection ratio (CMRR) at 60Hz was greater than 90dBu, while

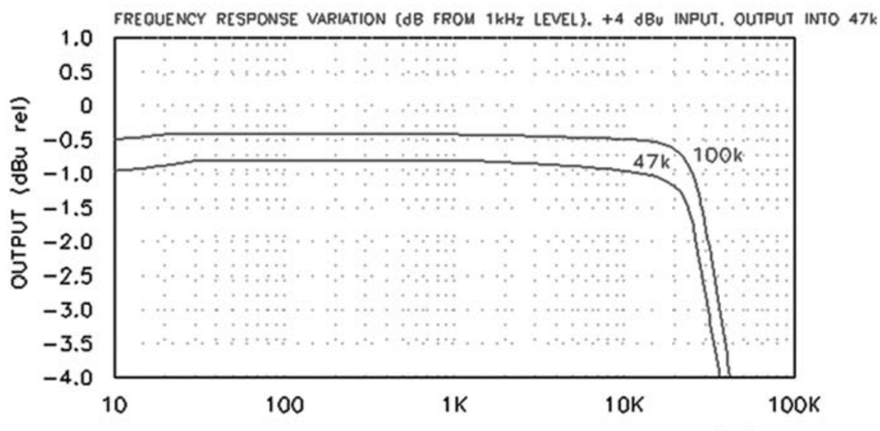

FIGURE 2: CI-2RR frequency response.

the CMRR at 3kHz was approximately 86dBu.

The phase shift with frequency is shown in **Fig. 3**, output with respect to input, referenced to 0dB at 1kHz. The source impedance is 600Ω and the output load is 47k. This graph is quite different from the "Deviation from Linear Phase" graph shown on the CI-2RR datasheet (available on the Jensen Transformer website).

Phase shift is not necessarily phase distortion. A totally benign time delay is represented in the phase domain as a phase shift that increases linearly with frequency; this is not distortion. Phase distortion occurs when there is a nonlinear relationship between frequency and phase. In other words, a time delay that changes with frequency[2]. Jensen specifies Deviation from Linear Phase (DLP) rather than phase shift, as the true measure of phase distortion.

Figure 4 shows THD+N versus frequency for the CI-2RR. The data from top to bottom at 20kHz represents 47k at +19dBu, 100k at +4dBu, 47k at +4dBu, and 10k at +4dBu. I engaged the test set 80kHz low-pass filter to limit the out-of-band noise.

In all instances the monitor output of the distortion test set, after the fundamental notch filter, showed the third harmonic, as you would expect of a magnetic core transformer. The CI-2RR THD+N is relatively insensitive to output loading. There is no dip in distortion at 60Hz, indicating the Jensen

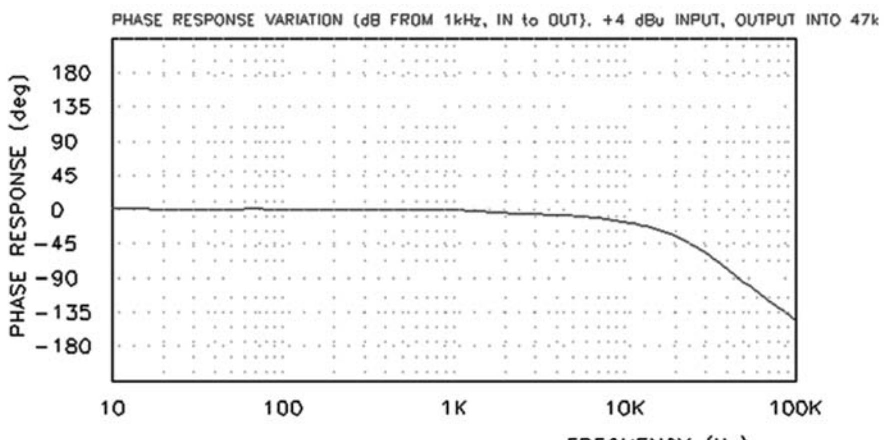

FIGURE 3: CI-2RR phase response.

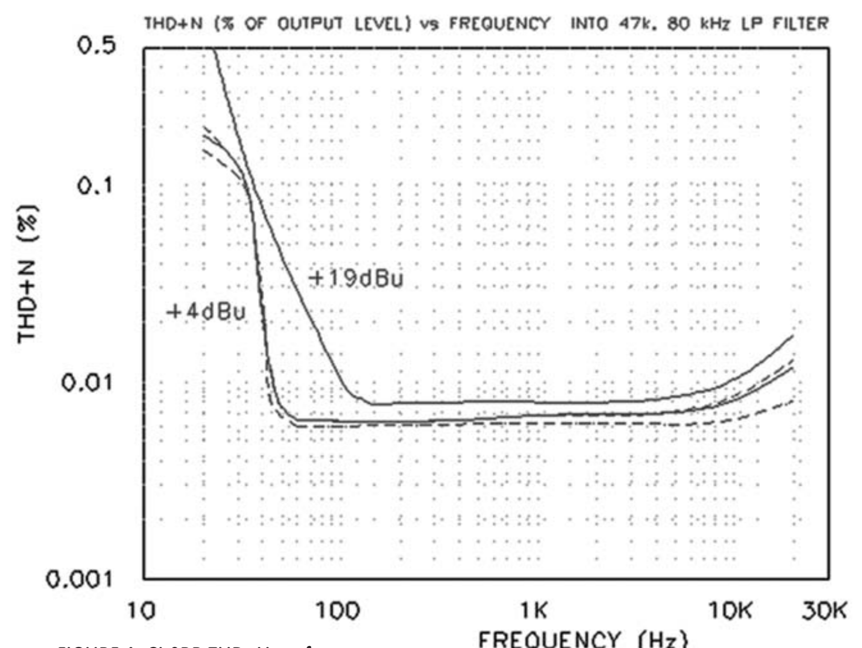

FIGURE 4: CI-2RR THD+N vs. frequency.

transformers are quite immune to external magnetic fields.

Figure 5 shows output THD+N versus input voltage into 47k at 20Hz, 1kHz, and 20kHz. The dashed line is for a 100k load at 1kHz. I engaged the test set 80kHz low-pass filter to limit the out-of-band noise. Using a 20Hz input signal, the CI-2RR endured 6.9V RMS (+19dBu) before "clipping" at 1% THD+N. If you want to convert the graph voltages to dBu, the formula is

dBu = 20*log(Vin/774.597)

I viewed the response of the CI-2RR to three square-wave test frequencies on an analog scope using a 47k load and 600Ω source impedance. The response at 40Hz was perfectly square up to 1Vpp output of my function generator, where some tilt made its appearance. The 1kHz square wave was also perfect until 2.5Vpp, where the tilt began. The 10kHz square wave did not begin to tilt until 13Vpp. This is because the saturation voltage of a magnetic core transformer is directly proportional to frequency (Faraday's Law).

I also found that the square-wave accuracy very much depends on the input impedance of the signal source, but not on the output load impedance. With a 50Ω source impedance, the 10kHz square wave had a slight peak on the leading edge. Increasing the source impedance to 1k while maintaining the same +4dBu input level caused the lead-

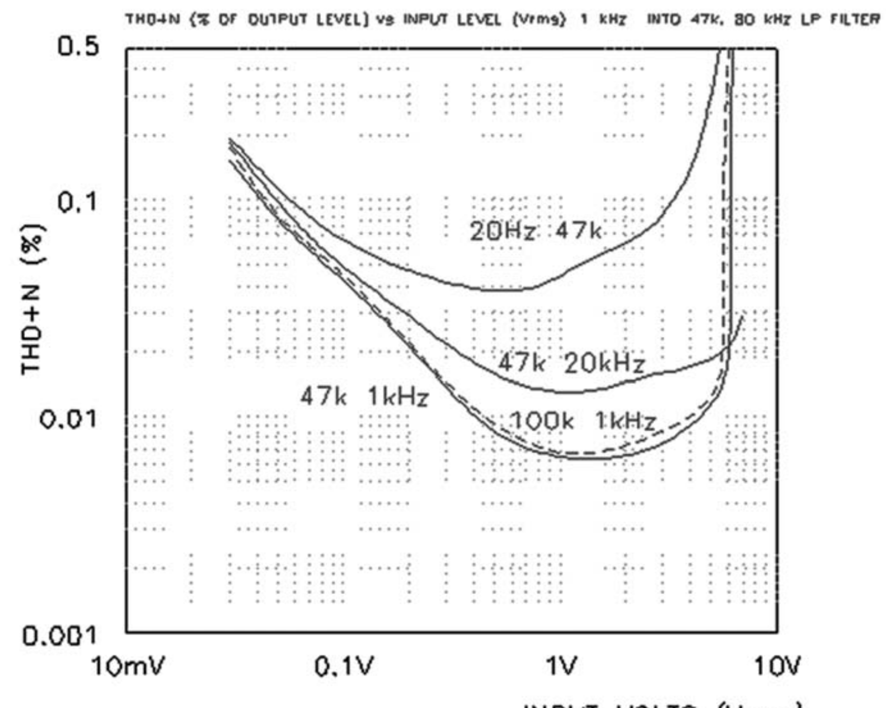

FIGURE 5: CI-2RR THD+N vs. input voltage.

TABLE 1: MEASURED PERFORMANCE.

Parameter	Manufacturer's Rating	Measured Results
Input Impedance, Zi, 1kHz, +4 dBu	47kΩ to 50kΩ	Approximately 48k
Insertion Loss, 1kHz, +4 dBu	1.0dB maximum	0.8dBu
Magnitude Response, ref 1kHz		
20Hz, +4dBu	–0.15dB minimum	–0.07dBu
20kHz, +4dBu	–1.0dB minimum	–0.31dBu
Deviation from Linear Phase (DLP)		Output Phase Shift:
20Hz to 20kHz, +4 dBu:	±2° maximum	+1° at 20Hz
		–35° at 20kHz
Total Harmonic Distortion (THD)		
1kHz, +4 dBu	<0.001% typ	.0064% (noise limited)
20kHz, +4 dBu	0.10% maximum	0.012%
Maximum 20Hz Input Level, 1% THD	+19dBu typ	+19dBu
Common Mode Rejection Ratio (CMRR)		
60Hz	95dB typ	> 90dB
3kHz	85dB typ	86dB
Output Impedance, Zo, 1kHz	4.65kΩ typ	Approximately 4k2
DC Resistance	2.26kΩ in, 1.90kΩ out typ.	2.18k in, 1.90k out
Capacitance, Zo, 1kHz, in to out	105pF typical	112pF L, 121pF R
Allowable Driving Source Impedance	0 to 2kΩ, 600Ω nominal	HF response sensitive (see text)
Allowable Load Impedance	10kΩto 47kΩ	Relatively insensitive 10k to 100k
Allowable Load Capacitance	0 to 100pF (cable + input)	N/A
Temperature Range	0° C to +70° C	N/A
Voltage Difference, input to output or any shield to case, 60Hz	24V RMS, 34V peak maximum	Not tested

ing edge of the 10kHz square wave to round over. The most accurate square-wave reproduction was at 600Ω. This indicates that the high-frequency response will be sensitive to the input source impedance.

Jensen President Bill Whitlock said the secondary R-C network was chosen to provide a two-pole Bessel high-frequency rolloff, which guarantees minimal DLP as evidenced by zero-overshoot and zero-ringing in square-wave response, within the specified source impedance and input voltage range.

Table 1 shows the Manufacturer's specifications and my measured results for comparison. *aX*

REFERENCES

1. "Product Review: Jensen Iso-Max Products," Charles Hansen and Ed Dell, *Audio Electronics*, 6/00, pp. 28-30.

2. "High-Frequency Phase Specifications—Useful or Misleading?," Deane Jensen, 1986 AES Paper 2398 (E-8).

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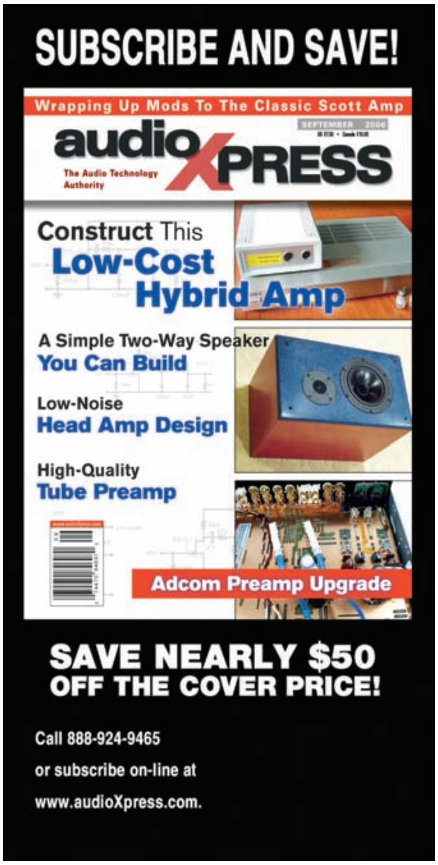

Ad Index

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METER READINGS

I like the fact that author Todd Sharp appreciates the utility of older high-quality audio measurement equipment ("A First Rate Audio Test Instrument, Part 1," June '06, p. 26). Readers should be aware of one limitation of the 403B. Like all the HP 400 series AC voltmeters, the 403B is not a true RMS reading meter despite the RMS VOLTS on the meter face. It responds to the average of the AC input waveform, but is calibrated to the RMS value of a pure sine wave. This could cause errors in measuring frequency response if the amplifier under test introduces significant distortion of the test signal.

The HP 3400A is a true RMS reading voltmeter with the same width and height as the HP 402C and 403B, but unfortunately it is too deep (279mm vs. 203mm) for the 3550B chassis.

Mr. Sharp has made valuable modifications to the units and I look forward to Part 2 of his project.

Chuck Hansen
Ocean, N.J.

Todd Sharp responds:

Thanks to Mr. Hansen for pointing this out. Indeed, the HP403b may indicate some degree of error while measuring waveforms with harmonic content. I have included an error chart from HP literature for consideration (**Table 1**).

Though I do appreciate the older analog gear (love those giant meters and knobs), I have personally found the HP 3400A to be considerably more difficult to repair and calibrate. Limitations notwithstanding, I stand by the 403b as a dandy little workhorse for mere pennies on the dollar! I also submit that its average responding scale may be relied on with a high degree of accuracy while viewing pure sine waves—which is most often the standard for bench testing basic amplifier performance.

Part 2 of this article (July '06) reveals that this modified set, which also features a convenient interface for oscilloscope and loudspeaker monitoring, provides the basic blocks for a compact yet affordable quality audio test bench that could well serve the neophyte, while offering portable test utility for the professional. I suspect that many readers would augment something like this with additional devices for more specialized purposes such as a good digital meter with a true RMS function.

After reading Mr. Hansen's letter, I did make some comparisons between *true* and *average* responding meters during the course of everyday repair work here at our shop. A few such comparisons follow:

While observing a sine wave just clipping a push-pull tube amp at 50W (THD measuring 5.4%), I noted that the modified 403b indicated identical amplitude as did a Fluke 8050A True RMS digital meter. The same test on a Hafler P1500 at 105W (1.2% THD) and frequency response tests on a Crown CE 2000 again produced identical readings.

Table 1. Effect of Harmonics on Model 403B Voltage Measurements.

Input Voltage Characteristics	True RMS Value	Value Indicated by 403B
Fundamental = 100	100	100
Fundamental + 10% 2nd harmonic	100.5	100
Fundamental + 20% 2nd harmonic	102	100 - 102
Funcamental + 50% 2nd harmonic	112	100 - 110
Fundamental + 10% 3rd harmonic	100.5	96 - 104
Fundamental + 20% 3rd harmonic	102	94 - 108
Fundamental + 50% 3rd harmonic	112	90 - 116

In regard to part 1 of Todd Sharp's test instrument article, most astute readers will have noticed that in Fig. 2 the symbol for transistor Q7 should be PNP, the arrow should point inwards, as it is the same transistor Q7 of Fig. 1.

Novice readers will have more problems with the two missing off-sheet connector symbols. One connects Q7's emitter from Fig. 2 to the circuit at the left of Q7's emitter in Fig. 1. The other connects the node comprised of resistors R41 and R2 along with C2's negative terminal from Fig. 2 to the node made of R41, C21's negative terminal and CR5's and CR7's anode terminals of Fig. 1.

This article, and others such as "Noise Meter Amp" (Jan. '05) and "Pooge for eBay Test Gear" (May '06)—both by Charles Hansen—should be of interest to all audio amateurs who want to improve their home laboratory with professional-quality test equipment. Professional solid-state test equipment built in the 1960s, 70s, and 80s is very useful to the hobbyist and amateur alike. Companies such as Hewlett-Packard (now Agilent), Tektronix, Fluke, General Radio (now Gen-Rad), Wavetek, and others sold many interesting instruments.

Most equipment designed for the professional market is now available for a fraction of their original price, sometimes even for free. They are mostly made up from discrete components (e.g., not surface-mounted) and use very few custom components, except for the cabinet, power transformer, switches, and analog meter movement. Most spare parts are commonly available. Service manuals are also available, sometimes for free, at many websites such as BAMA http://bama.sbc.edu/. To find such test equipment look at yard sales, HAMfests, surplus and second-hand stores, pawn shops, used test equipment stores, and at online auction sites.

The problems you may encounter: cleaning the outsides and insides, troubleshooting and repair, paying more for the manual than what you paid for the instrument, looking for parts made of unobtainium (a fake element name to mean that some parts are no longer available), buying more than one unit so you can have spare parts, and generally learning and having fun with electronics.

One instrument of interest is the Hewlett-Packard 4328A Milliohmmeter, which was designed in the late 1960s. This analog instrument offers AC resistance measurement, 1mΩ to 100Ω full-scale in 11 ranges, low test signal 20mV peak across the sample, typical 200μV peak, 1kHz test frequency, and the possibility of measuring all this with 150V DC across the sample to be tested. Accuracy is 2% even when the reactance of the sample is twice the full scale range reading, so you can use it to measure capacitor internal resistance (ESR) quite easily. It was available with option 001, a rechargeable battery. This model option will benefit from Mr. Sharp's modifications to replace the batteries. Armed with a capacitance meter and this milliohmmeter, you can track down old dried-up aluminum electrolytic capacitors very quickly.

Daniel Dufresne
Saint-Laurent, Québec

Todd Sharp responds:

I thank Mr. Dufresne for catching the error in Fig. 2, where Q7 is shown as a NPN transistor. It should be a PNP. My apologies for not catching this in proofreading.

Regarding the other questions raised: Readers may notice the vertical dotted line in Fig. 2 which depicts "existing circuit" and "new circuit." The emitter of Q7 is never disconnected, although I can see where you might draw that conclusion based on the way Fig. 2 is presented. The text of the article explains this along with the other connection in question.

I am pleased that you found the article of interest.

I enjoyed Todd Sharp's article on the HP 203c renovation project. I really think a person could get the new NiCads specially made up at Battery Plus instead of putting capacitors and zeners in their place and changing the circuits.

I need help on a JL Audio 1000/1 bass amp. The problem is that the green power LED lights (normal), and the yellow LED lights (low ohm) with the speaker disconnected (on purpose in case there was a voltage, no speaker damage would occur). There are no shorts that I can find in the 31 N-channel power MOSFETs or the four dual rectifiers that look like TO-220 transistors, and no shorts between the speaker terminals, which measure about 2MΩ.

Please tell me this information if possible. This amp is Class D fed with pulse width modulation. I don't know what else to test.

Tim Rasch
tim5310@yahoo.com

Todd Sharp responds:

You make a good point in your letter. I was not aware that you could have NiCads made up in custom configurations. In either case, I would consider the cost vs. application carefully before making the investment. There are advantages and disadvantages to rechargeable battery operation. As the article points out, the NiCad cells are often the problem area in resurrecting not only the 403b and 204c, but a wide variety of test gear from this era. Such gear is otherwise of top quality and readily obtainable at bargain prices.

I am no expert on battery technology, but I will share my limited experience on the topic since you have raised it. The advantages of battery-operated test equipment are mainly twofold: 1) AC line isolation and freedom from ground loops, which can be especially troublesome at very low measurement levels, 2) remote operation without having to plug in to a power source, provided the cells are adequately charged.

The disadvantages as I have experienced them are: NiCad cells have a limited charge cycle life and are susceptible to memory effect. If you do not discharge them completely before recharging them, they tend to yield a much shorter service life. Therefore, you will be buying more batteries at some point in order to keep your stuff operational.

There are a few methods of resurrecting NiCads such as "zapping" with high voltage pulses, but this is not possible with 6V types. As I understand it, all batteries are actually 1.2V cells. The 6V types required in this set are made up of five 1.2 cells stacked in series and then sealed as a single 6V cell. Each HP piece in my article requires four 6V cells. Personally, I have rare cause to use the "battery power" function. I just leave them plugged in, which means I would be buying new custom made NiCads more often than if I allowed the cells to discharge fully.

You also need to be aware that these cells are required to be in good shape even when operating from the AC line, because they are in fact utilized as filters for the rectified AC line supply. As if that isn't enough, if one of the cells (in the string of 20) is going bad (which is not uncommon), the resulting leakage causes further power supply trouble.

So, replacing these buggers with a few filter caps and a few diodes may make more sense than you would initially imagine. There are newer technology NIMH cells available which seem to work much better than NiCads. Perhaps someone with deeper understanding of current rechargeable battery technology could pick this topic up and shed more light for all of us.

Thank you for your comment. Glad you found the project of interest.

CORRECTION

Readers should note that in part 2 of my article (July '06), J2 (speaker) is drawn incorrectly in Fig. 2 (p. 42) as a shorting type jack. It should be a two-terminal mono jack without shorting contacts. It is listed correctly in the parts list as a Switchcraft #11.

There are also two typos in Fig 2: SW3 position "X11" should say "X1" and the two + binding posts should be indicated as BP3 and BP4.

Todd Sharp
todd@amprepair.com

NOISE AND DISTORTION

The Audio Amateur has published noise values for some operational amplifiers—in 1/1979 by Michael Lampyon and Don Zukauckas as well as in my 2/1994 article. These noise values are not frequency-weighted.

Due to the Fletcher-Munson effect, the human ear is most sensitive to noise in the frequency range 900Hz-7kHz. Noise from operational amplifiers should be filtered to match the Fletcher-Munson curve. An "A"-weighting filter was published by Phillipe Trolliet in *TAA* 3/1981.

If you use the operational amplifier in an RIAA amplifier, then you must sum the effects on the noise of the RIAA filter and the weighting filter. I use a very simple filter for noise measurements (**Fig. 1**).

The distortion values for some of the operational amplifiers are taken from my article in *TAA* 2/94. Filter

FREQUENCY	A-FILTER	A-FILTER + RIAA	ACTUAL FILTER
20Hz	-50dB	-31dB	-28.5dB
50Hz	-30dB	-13dB	-21dB
100Hz	-20dB	-7dB	-16dB
500Hz	-3.5dB	-0.5dB	-3dB
1kHz	0dB	0dB	-2.5dB
5kHz	0dB	-8dB	-10dB
10kHz	-3dB	-17dB	-20dB
15kHz	-6dB	-23dB	-24dB

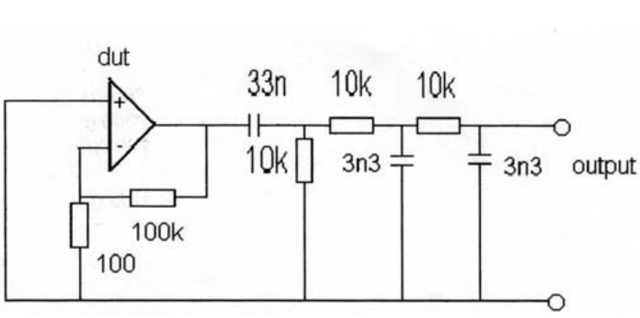
OPERATIONAL AMPLIFIERS

SINGLE	NOISE MV	DISTORTION %
OP 07	140	0.53
OP27	-	0.0016
OPA27	80	-
OP37	-	<0.0003
TL061	860	0.27
TL071	433	0.0008
TL081	490	0.0019
OPA121	155	0.39
OPA134	238	-
OPA137	792	-
OPA177	116	-
LM308	530	0.13
LM318	420	<0.0003
LF351	528	0.0042
LF356	655	0.032
LF357	457	0.0016
OPA602	330	-
OPA603	103	-
OPA606	35	-
AD645	187	-
LM725CN	71	-
741	-	0.78
LT1001	149	-
LT1016	11	-
LT1115	45	<0.0003
TLE2061	952	-
2134	89	0.0003
CA3140	808	0.0045
NE5534	113	-
LM6181	158	-
DUAL		
TL052	480	-
TL062	690	-
TL072	445	0.0010
OP80G	104	-
TL082	445	0.0004
LF353	440	0.0004
LM358	990	0.0004
TIS512	251	-
UA772	490	0.0032
LM833	131	-
1458	920	0.41
2139	103	0.0003
LM2904	635	-
CA3240	785	0.0039
4558	325	0.0078
NE5512	735	0.016
NE5532	185	<0.0003
QUADS		
TL074	515	0.029
TL084	595	0.029
LM324	655	0.33
LF347	610	0.0005
LM348	905	0.44
LM349	1590	0.0026
OP400	135	-
LT1014	460	1.2
MC3403	1010	0.52
NE5514	905	0.018

characteristics are for "A"-filter, "A"-filter + RIAA, and the filter that I used.

Rickard Berglund
Sweden

FIGURE 1: Noise measurements in op-amps.



KEEPING COOL

I read with interest part 2 of Dennis Colin's Mad Katy 250W tube power amp (July '06 *aX*). I see that an "Optional" fan is included. I suggest that the fan is not optional but a necessity. The measurements as given make the centers of the KT88s 3.14″ apart. (I must say further apart than some designs I have seen of late.)

The M-O Valve Co Ltd Bulletin on the GEC KT88s December 1974 states that KT88s should be 4″ apart center to center, and correctly oriented. The reason for this is that with KT88s producing this amount of power the envelope/bulb temperature gets up to about 250° C, which is 2.5 times the boiling point of water. With so much heat concentrated in such a small area, I question the reliability of surrounding components.

I have recently built a 100W per channel tube amp using KT88s with the M-O Valve specs and have put in a fan to do some cooling.

Alan Smith
jbandaj@clear.net.nz

Dennis Colin responds:

As the article shows (Table 2, p. 55), at +24° C ambient and no signal (398W AC consumption), the hottest KT88 envelope spot is 195° C (no fan) and 163° C with fan. Calculating for maximum music signal (≈ 500W AC line) and 100° F (38° C) ambient, the KT88 envelope would reach about 225° C (no fan) and 190° C with fan.

This is with the Papst 3412L fan (12V design) operated at 6V. I measured its wind speed as 4.6mph (2.06m/s) and its "A" weighted noise as only 7dB SPL at 3m. Running the fan at 12V produces a wind speed of 10.3mph (4.60m/s) and noise of 24dB SPL at 3m. This would more than double the cooling.

I power the fan from the 6.3V DC supply in the amp. You could also connect it across the ±6.3V supplies for 12.6V operation. The higher noise of 24dB SPL at 3m is just barely audible here in the New Hampshire mountains on a winter night.

I appreciate your letter and agree with you. I always use the fan (on 6.3V). Incidentally, the reason the KT88 maximum envelope temperature is 195° C, not 250°, is that the plate plus screen dissipation is only 28W maximum, with reference to the recommended Genelex rating of 40W.

HIGH-QUALITY AC PLUGS, SOCKETS

A few weeks ago I rewired an auxiliary power supply with audiophile-approved AC wire and a robust 15A plug, replacing one of "lamp cord" quality.

By chance I stumbled across a Marinco plug (marinco.com), the most wonderfully engineered and beautifully made plug I've used. Previous plugs have been the type that look good enough, are reasonably priced, and made in China. I am sure their retailers are making handsome profits on them.

The Marinco plugs and wall outlet sockets are identified as being made in the USA, and in fact cost no more than the imports. I have therefore won on all three fronts—quality American manufacture, competitive pricing, and unmistakable sonic improvements.

Darcy E. Staggs
Orange, Calif.

PHASE INVERTER DESIGN

Robert Bennett's article, "An Improved Split-Load Phase Inverter" (*aX* 7/06), was interesting, but left me puzzled as to his interpretation of circuit action.

I agree that if the input to a phase splitter is truly made between its grid and cathode, then the undesirable features he mentions would be eliminated. That's because such a connection would eliminate the negative feedback inherent in the stage when it's used in the classic configuration (Fig. 1, p. 25 of July issue). Unfortunately, despite his explanation, I don't see how his approach provides that connection in either example he's offered (Figs. 2 and 3, pp. 26 and 27).

A sure-fire way to produce it would be to use transformer coupling into the splitter, with the secondary connected in place of his R12 of Fig. 2. Of course, C7 and C8 would be eliminated, and R5 would be lowered to match R6 to balance the outputs. In this setting, the raw stage (not counting any transformer action) would actually provide some gain (about 8 to either output, or 16 between both outputs with a 6J5), and behave similarly on each output with respect to frequency balance, output impedance, and loading, all as Mr. Bennett suggests.

The transformer coupling in my example truly allows the inverter's input to be "returned" to the top of R5 rather than ground, so that the signal applied from the transformer between the grid and cathode is unaffected by the cathode signal appearing between the top of R5 and ground. In other words, the input signal "floats" on top of the cathode output signal.

In Mr. Bennett's example, however, the signal from the pentode stage is still effectively applied between the inverter's grid and ground, because that is the point the cathodes of both stages reference. On paper it may *look* as though the signal is applied between the inverter's grid and cathode, but I suggest that in reality, it's not.

Consider in his Fig. 2 if you remove C8 for a moment, and reduce R5 to match R6 at 20K. Then, if a 10V signal (ref. ground) is produced at the plate of V1, it means that 92.6% of it is produced across R3, with the remainder across R11 if the B+ line is properly bypassed. Also about 95% of V1's plate signal will appear across R5, because the inverter is now operating in a classic configuration with the cathode output signal applied as NFB against the inverter's input signal.

However, if you now re-install C8, and increase R5 to the listed value of 40K, the circuit action changes, but not as I believe Mr. Bennett suggests. Now, a 10V signal at the plate of V1 will still produce about 95% of that voltage across R5, *but also* across R11, which means only 5% of the 10V drive signal is now appearing across R3, versus nearly 93% before; hence, the bootstrap action.

In this case, the effect of the bootstrap is to practically make R3 dynamically disappear. Mr. Bennett states that such treatment has no effect on the gain of the pentode. I question this as well. The bootstrap has effectively made R3 "appear" as almost 19 times its actual value, which allows the pentode to approach its theoretical maximum gain by operating in a nearly constant-current mode. I believe this is the real benefit of the design. That is, it maximizes the gain of the pentode stage, rather than enhancing the performance of the inverter stage. In fact, I believe the performance of the actual inverter as shown in Figs. 2 or 3 has pretty much the same drawbacks as listed at the beginning of the article, but now with two new concerns.

As shown, any adverse loading on the cathode output will not only have a potentially opposite effect on the plate output as before, but can now also upset the bootstrap connection as well. This could lower the pentode's gain and reduce the effectiveness of any global feedback network in place. Additionally, because R5 works in tandem with R11 to provide a balance with R6, the value of R5 is doubled over that of R6. While this restores AC balance, it also reduces the DC current that the stage can pass (as opposed to that where R5 = R6), which in turn reduces its peak output capabilities. You should also consider this if you're contemplating a change to the bootstrap connection.

Finally, Mr. Bennett's comments on heater bias for a splitter design are well founded, and often neglected. I would also add that the bias supply for the heaters should always be adequately bypassed to ground as a further aid in minimizing noise.

All of that said, the phase splitter is a good circuit, and there is much to commend it for. Most of its limitations can be effectively dealt with, because its well-publicized drawbacks rarely produce the level of problems in reality that it's been debated to have in theory over the last 50+ years. I would appreciate any comments Mr. Bennett might have on this in case I just don't get it—which, of course, is always possible!

David C. Gillespie
Atlanta, Ga.

Robert Bennett responds:
David Gillespie's letter raised some interesting points.

His suggestion of a split-load phase inverter, with input transformer-coupled between grid and cathode, is an intriguing one, which might well have practical merit.

As far as the bootstrap circuit is concerned, you can analyze it in a number of ways, but the analysis requires some care, considering the circuit as a whole, without separating the parts. Surprisingly, the gain and output resistances of this circuit are very similar to Mr. Gillespie's suggested transformer-coupled circuit, provided that the pentode plate resistance is much larger than its load resistance.

The voltage at the 6J5 cathode output is about 90% of the grid voltage. The bootstrap connection increases the effective anode load of the 6SJ7 by a factor of 1/(1-0.9) = 9 roughly.

The 6SJ7 has a plate resistance of about 3MΩ and a transconductance of about 1mA/V under the circuit conditions. The higher effective load increases its gain by about six times, from 230 to 1200. The total gain from input to one output is about 1100.

The split-load phase inverter normally has a very low impedance at the cathode output (roughly 1500Ω for the 6J5) because of the negative feedback from the cathode resistor. The impedance at the anode output is much higher.

The output impedance at the cathode is: (6J5 plate resistance + cathode resistor)/(mu+1) = 1500Ω.

However, in the bootstrap circuit, there is an equal amount of positive feedback increasing the pentode's gain. This largely cancels out the effect of the negative feedback, increasing the cathode output impedance back to a fairly normal value.

The bootstrap increases the effective output impedance of the cathode channel by a factor of about 6, which brings it up to roughly the same level as the output impedance of the anode channel.

With a normal split-load phase inverter, if the cathode output is shorted, the output at the anode increases dramatically, by a factor of about 15 for a 6J5, for example.

With the bootstrap circuit, shorting the cathode output also shorts the bootstrap feedback, reducing the pentode's gain by a factor of about 6, so the output from the triode increases by a factor of only about 2.

Mr. Gillespie considers the effect of a voltage generator applied between pentode anode and earth. This would short out the bootstrap feedback and lead to erroneous conclusions. It would be better to consider the pentode as a current generator in parallel with a very high resistance (the plate resistance of the pentode).

Mr. Gillespie also comments that the maximum output swing of the bootstrap circuit is reduced because of the AC load of the bootstrap. This is correct. This problem could be avoided by direct-coupling the top end of the pentode load resistor to the triode cathode. Mr. Gillespie's final point about having the heater supply well bypassed is particularly important with high impedance circuits such as this one.

I agree that the traditional form of split-load phase inverter works well enough in practice if the amplifier is not overdriven. Perhaps if other circuits had been used for as many years, they would also have been found to have drawbacks. Finally, I would again like to thank Mr. Gillespie for his letter, which has caused me to look again at the references and do some hard (but worthwhile) wrestling with circuit analysis. *aX*

What's New on the aX website?

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Plate Curves from Charles Hansen's "Renovating the Scott 222C, Part 5: Output Tube Alternatives to the 1789 and Tube Matching Tests," *aX* 9/06.

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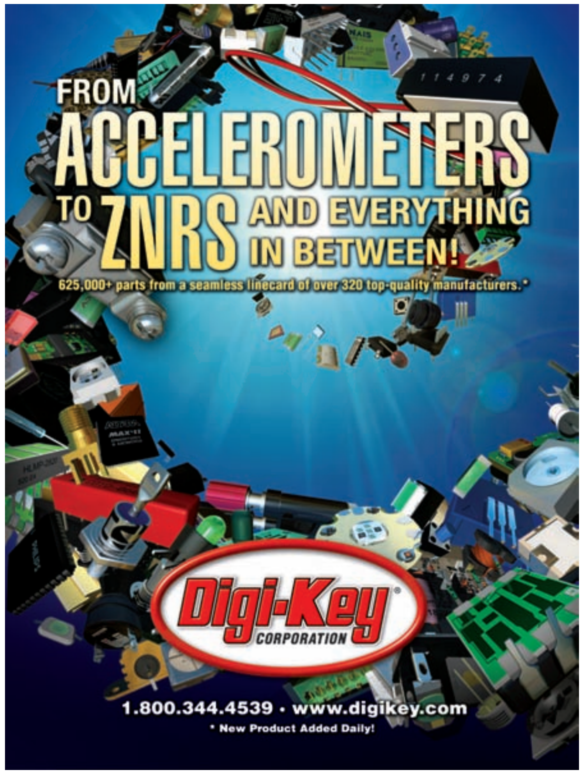


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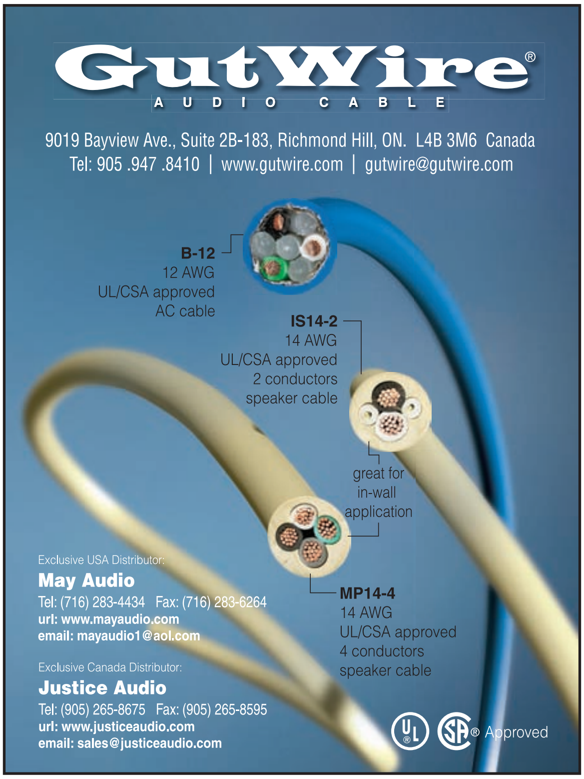
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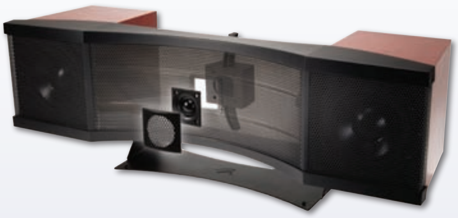

The M3 Silicon Valve solid-state microphone, from **MXL Microphones**, is a large diaphragm, pressure-gradient condenser mike. The M3 does not require an external power supply, and does not waste time warming up like tube microphones. There is another version, the M3-P, voiced specifically for female vocals. For additional information, see www.mxlmics.com.

Impact Acoustics introduced a new line of SonicWave Pro-Audio Cables, featuring gold-plated pins, textured grip, and precise soldering techniques that eliminate the negative effects of dissimilar metals. The coaxial audio cables feature 75Ω geometry and oxygen free conductors. There are also twisted pair, shielded audio cables, which are available in XLR male to female and ¼″ TRS male to male. You can find these at www.impactacoustics.com.

Equi=Tech announced the Model 20WQ Wall Cabinet System with twice the power of its smaller versions in the same amount of space. With a 200A capacity and room for 20 circuits for hard wiring, the 20WQ features the Q type transformer that keeps current and voltage accurately in phase. The 20WQ also contains a lower cabinet which houses the toroid transformer that drives the system, and there is additional EMI/RFI filtering available. To learn more, go to www.equitech.com.

The InstaSnake, from **Energy Transformation Systems, Inc.,** consists of two small boxes with four XLR microphone inputs and a shielded RJ45 jack. It accepts mike line level, analog or digital audio, and supports phantom power. For more information, contact tkayne@etslan.com or visit www.etslan.com.

New from **TDL Technology** is the Model 444 Stereo Headphone Amplifier, featuring wide bandwidth, low noise, and DC operated balance and volume. Minimizing unwanted signal pickup, the 444 works with virtually all types of headphones. The 444 has flat frequency response, comes with a 24V DC wall power supply, and has 47k input resistance for each stereo channel. Learn more at www.tdl-tech.com.

The TA-500, a three-way, full-range enclosure from **Turbosound**, provides wider dispersion characterstics. Initially available in a fixed install/ground stacked variant, the TA-500 is another in the Aspect product range. To find out more, please go to www.turbosound.com.

Danley Sound Labs introduced the SH-100B loudspeaker, with frequency response to below 40Hz and maximum SPL in the upper 120dB range. Compact at 29.5″ × 20.5″ × 19″, the SH-100B is highly mobile and can be used in houses of worship, performing arts centers, live music venues, and more. Visit www.danleysoundlabs.com for more information. *aX*